AN UNOFFICIAL GUIDE FOR BEGINNERS

Andy Wehmeyer on Car Audio, Explained for Beginners

What one of car audio's most experienced designers teaches about speakers, installation and tuning, with every point explained in plain English.

Paraphrased from Andy Wehmeyer's public forum posts, blog, tuning guides, videos and podcasts, with a plain-English explanation of every point. Not written, reviewed or endorsed by Andy Wehmeyer, Audiofrog, Harman or JBL.

Every point links to its sources. Built 13 October 2026.

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Please read first: what this guide is, and what it is not

This is an **unofficial** guide. It was compiled from Andy Wehmeyer's public forum posts, blog, tuning guides, videos and podcasts, dated 2008 to 2023 (plus one livestream from January 2024). Andy Wehmeyer is a co-founder of Audiofrog and was the Harman product manager behind the JBL MS-8 processor.

- **His points are paraphrased, not quoted.** Each point is a summary in the compiler's words of what he said. Only words inside quotation marks are his own, and those quotes are kept short.
- **He did not write, review or endorse this guide.** Neither did Audiofrog, Harman or JBL. Any mistakes are the compiler's, not his.
- **Check the sources for his exact words.** Every point ends with a "Sources:" line linking to the forum post, blog article, guide or video it came from. Where it matters, follow the link and read what he actually said, in context.
- **The "In plain English" explanations are the compiler's.** They are written to explain his meaning to a beginner, not to add to it. When an explanation gives general background (what a tweeter or a decibel is, for example), it is worded as general background ("A crossover is..."), not as his opinion ("He says...").
- **Product names belong to their owners.** Audiofrog, JBL, Harman, MS-8, Logic 7, Bose, REW and every other product or company name mentioned belong to their respective owners. Naming a product is not a recommendation by the compiler.
- **He sells much of what he recommends.** Weigh his comments on his own products accordingly.
- **Car electrical and installation work can damage a car or injure you.** This guide explains ideas; it is not installation instructions. When in doubt, use a qualified installer.

How to use this guide

How each point is laid out

Every point appears in two parts:

1. **His point.** A bold one-line summary, then a short paraphrase of his explanation, any tags (see below), and a "Sources:" line with links. This part is copied word for word from the compiled list of his points, so it stays faithful to him.
2. **In plain English.** A short explanation of the same point for someone who has never installed a speaker: what the words mean, what he is saying, and, where he gave one, what it means in practice.

If the two ever seem to disagree, trust the first part and its sources.

What the reference numbers mean

Points are numbered by chapter, subsection and position. "**3.9 #4**" means chapter 3, subsection 3.9, the 4th point in that subsection. A few references point to a named list inside a subsection: "3.1 evolution" is the "How his procedure evolved" list inside subsection 3.1.

The chapters are:

1. **Sound and speaker placement:** how stereo works in a car, reflections, how speakers spread sound, where to put them, center channels, rear speakers and upmixing.
2. **Installation and hardware:** doors and midbass, sound deadening, pods, tweeter mounting, choosing drivers, tweeter protection and wiring.
3. **Tuning and measurement:** his step-by-step process, measuring, target curves, EQ, crossovers, phase, delay and the JBL MS-8.
4. **Bass, power and factory systems:** how the car boosts bass, blending the subwoofer, boxes, amplifier power, setting gains, noise and upgrading factory systems.
5. **Myths, buying and industry:** common myths, reading specifications, buying, choosing a shop, the business, competitions, listening, and his background.

What the tags mean

- `RARE`: the point goes against, or well beyond, common car-audio advice. The explanation often says what the common belief is.
- `Changed view: ...`: he changed this advice over the years. The tag gives the earlier and later positions with dates. The guide explains both and does not pick a side.
- `Conflict: ...`: his sources say different things, for example a tuning guide and a forum post, or his own words and his company's sales copy. Again, both sides are explained without choosing.
- `Era: ...`: the advice belongs to a particular period or product, such as his years at Harman or the JBL MS-8 processor. Products and software change, so check whether it still applies.

What the source labels mean

Each "Sources:" line uses short labels. The most common are:

- **DIYMA 2021-09-14** (a date): a post on the DIYMobileAudio forum (diymobileaudio.com) on that date. The forum renamed his two accounts to the numbers **"14642"** (posts from 2008 to 2016) and **"156546"** (2017 to 2022). Posts under both numbers are his.
- **DIYMA MS-8 FAQ:** his MS-8 question-and-answer thread on the same forum, from his Harman years.
- **DIYMA gain guide (attributed):** a gain-setting guide on the forum that is attributed to him; it is not confirmed as his own writing.
- **Audiofrog blog** (with a month and year): his signed Tech Blog articles on the Audiofrog website (December 2020 to October 2021).
- **Tuning PDF v1 / v2:** his free Audiofrog tuning guide, first version about 2018, second about 2023.
- **COMPANY:** an unsigned Audiofrog web page. It may reflect his views, but because it is not signed, it is not certain that he wrote it.
- **A-Series manual, MS-8 manual:** product manuals (an Audiofrog amplifier manual; the JBL MS-8 manual).
- **OFCAP Ep. 19 @1:45:06:** the Old Fashioned Car Audio Podcast, episode 19, at 1 hour 45 minutes 6 seconds. The link jumps to that moment.
- **SQOLOGY:** a car-audio podcast from 2017–18, with the part number.
- **CMA** or **CMA Networks** (with a year and timestamp): CMA Networks videos from 2021 to 2023. **CMA Connected** is its October 2021 livestream.
- **B1** and **B2:** two livestreams: B1 is the Absolute Electronix livestream (March 2023) and B2 is the CMA Connected livestream (October 2021).
- **D&F live:** Dean and Fernando's livestream (January 2024).
- **EDUCAR:** a technical tip published in EDUCAR magazine.
- Other labels name the publication or site directly, such as a magazine, a retailer page, a product listing or a third-party measurement review.

Video and podcast points were taken from automatic captions, so timestamps are approximate and wording may be slightly off. Follow the link and listen if the exact words matter.

A suggested reading order for beginners

1. **"Car audio in 10 minutes"** (next in this guide). It gives you the basic words and ideas the rest assumes.
2. **"Start here"**: 50 of his most distinctive points, each with a plain-English note. It is the quickest way to see how he thinks.
3. **Chapter 5 (Myths, buying and industry)**, if you are about to buy a system or choose a shop.

4. **Chapter 1 (Sound and speaker placement)**, which explains why a car sounds the way it does. Most of the later advice builds on it.
5. **Chapter 2 (Installation and hardware)**, then **Chapter 4 (Bass, power and factory systems)**.
6. **Chapter 3 (Tuning and measurement)** last. It is the most technical and makes most sense once the rest is familiar.

Keep the **glossary** at the back open as you read. Any term in bold or in quotation marks that you don't recognise is probably in it.

Car audio in 10 minutes

This primer is general background, written by the compiler for complete beginners. It is not a summary of Andy Wehmeyer's views; his views start in the next section.

The parts of a system

Think of a car audio system as a chain. The music passes along it from one link to the next, and the weakest link limits the result.

The source (head unit). This is the radio or screen in the dash, or a phone feeding it. It produces a small electrical signal, measured in volts, that carries the music. Its volume knob controls how strong that signal is. In many modern cars the factory head unit is tied into the car's other systems and is hard to replace.

The processor (DSP). A digital signal processor, or DSP, is a small computer for sound. It takes the music signal, splits it into separate channels for each speaker, and adjusts each one. It can change tone (equalization, or EQ), decide which speaker plays which notes (crossovers), hold back one speaker's sound by a few thousandths of a second (delay), and set each channel's level. Many factory systems already contain one, often hidden inside the factory amplifier.

The amplifiers. The signal from the source or DSP is far too weak to move a speaker. An amplifier ("amp") makes it powerful enough. Amplifier power is measured in watts (W). An amp has a "gain" control: it does not set power, it matches the amp to the strength of the incoming signal. If any part of the chain is pushed beyond its limit, it "clips": the tops of the sound waves are cut off flat, which sounds harsh and can damage speakers.

The speakers (drivers). A speaker turns electrical signals back into sound by moving a cone or dome back and forth. No single speaker can play every note well, so systems divide the work:

- **Tweeter:** a small dome, often about 1 inch (25 mm) across, that plays the highest notes, such as cymbals and the "s" sounds in voices.
- **Midrange:** a small cone, roughly 2 to 4 inches (50–100 mm), that plays the middle notes where most of a voice lives.
- **Midbass (or woofer):** a larger cone, often 6 to 8 inches (150–200 mm) and usually in the door, that plays upper bass and lower middle notes, such as the body of a kick drum or a male voice.
- **Subwoofer:** a large cone, often 8 to 15 inches (200–380 mm), in its own box, that plays the deepest bass.

A system with a tweeter and a midbass on each side is a "2-way". Add a midrange and it is a "3-way". The individual speakers are often called "drivers".

Crossovers. A crossover splits the music so each speaker gets only the notes it handles well. A "high-pass" filter lets high notes through and blocks low ones; a "low-pass" filter does the opposite. The point where one speaker hands over to the next is the "crossover frequency". A **passive** crossover is a small box of electrical parts (capacitors, coils and resistors) between the

amp and the speakers; it is simple, but fixed. An **active** crossover sits before the amplifiers, usually inside a DSP, and gives each speaker its own amp channel, so every setting can be adjusted and fine-tuned.

Wiring. Power cables bring current from the car's battery to the amps. Signal cables (often "RCA" cables, named after their plug) carry the music from the source or DSP to the amps. Speaker wires carry the amplified signal to the speakers. Good connections, and correct polarity (plus to plus, minus to minus), matter: swap the two wires on one speaker and it moves backwards relative to the others.

The basic units

Hertz (Hz) measures frequency, the pitch of a sound: how many times per second the air vibrates. Low numbers are low notes. Human hearing runs from about 20 Hz to 20,000 Hz. **Kilohertz (kHz)** is a thousand hertz, so 20 kHz is 20,000 Hz. As a rough map: deep bass sits below about 80 Hz, midbass roughly 80–300 Hz, the midrange from a few hundred hertz to a few kilohertz, and treble above that.

An octave is a doubling of frequency: 100 Hz to 200 Hz is one octave, and so is 1,000 Hz to 2,000 Hz. Hearing works in octaves, so audio graphs usually space them evenly.

Wavelength is the physical length of one cycle of a sound wave. Low notes have long waves (about 4.3 m, or 14 ft, at 80 Hz); high notes have short ones (about 3.4 cm, or 1.3 inches, at 10 kHz). This is why the size of a car, and the distances between speakers and surfaces, matter so much.

The decibel (dB) measures differences in level (loudness). It works on ratios, not plain steps. An increase of about 1 dB is barely noticeable, 3 dB is clearly noticeable, 6 dB is a big change, and about 10 dB sounds roughly "twice as loud". Each extra 3 dB of output needs about twice the amplifier power, so 10 dB more needs about ten times the power. "dB SPL" is sound pressure level, the actual loudness measured in the air.

The millisecond (ms) is a thousandth of a second. Sound travels about 343 m (roughly 1,125 ft) per second in air, so in 1 ms it covers about 34 cm, or about 13.5 inches. In a car, where speakers sit at different distances from you, differences of a millisecond or two decide which speaker's sound reaches you first.

Phase describes where a wave is in its cycle, measured in degrees (°). When two speakers play the same note "in phase", their waves rise and fall together and add up. When they are 180° apart, one pushes while the other pulls, and they cancel.

What listeners mean by imaging, staging and tonal balance

Imaging is the illusion that a singer or instrument sits at a precise spot in front of you, even between the speakers, where there is no speaker at all. Good imaging puts a lead vocal in the middle of the dashboard, solid and small, as if someone were standing there.

Staging (the "soundstage") is the overall picture: how wide the band seems, how high (at dash level or down at your knees), and how far away. A good stage sounds as if the music is spread across the front of the car, in front of you, rather than coming from the door by your knee.

Tonal balance is how the low, middle and high notes sit against each other. Too much bass sounds boomy, too much treble sounds harsh, and a weak midrange makes voices thin. Tuners often aim for a "target curve", a chosen shape for the balance from low to high notes, measured with a microphone.

What tuning is

Tuning is adjusting the processor so the system sounds right from the listening seat. It usually involves four kinds of setting:

- **Crossovers:** which notes each speaker plays, and how sharply it hands over to the next.
- **EQ (equalization):** turning particular frequency bands up or down to correct the balance. A "parametric" EQ lets you choose the center frequency, the amount, and the width (often called "Q": high Q is narrow, low Q is broad).
- **Delay, or time alignment:** holding back the nearer speakers by a millisecond or so (a few thousandths of a second at most) so their sound arrives with the farther ones.
- **Levels:** how loud each speaker or side is relative to the others.

Tuners measure with a microphone and software. A common tool is an **RTA** (real-time analyzer), which shows the level at each frequency while a test signal plays, often **pink noise**, a hiss with equal energy in every octave. Another common approach is a **sweep**: a tone that glides from low to high, analyzed by software such as REW (Room EQ Wizard). The result is a "frequency response" graph: frequency from low to high along the bottom, level in decibels up the side.

Why a car is a hard place for sound

Home speakers are usually placed in front of you, at equal distances, in a room several metres across. A car breaks almost every one of those conditions.

- **You sit off-center.** The driver is much closer to the left speakers than the right (the reverse in right-hand-drive cars). The near side's sound arrives first and louder, which pulls the image toward it.
- **There are two (or more) seats.** A fix that suits the driver's seat can make the passenger's seat worse.
- **Hard surfaces are very close.** Glass, the dash and plastic panels sit centimetres from the speakers. Sound bounces off them and arrives a fraction of a millisecond after the direct sound. Where the bounce and the original are out of step, they cancel at some frequencies and add at others, a pattern called **comb filtering**, because on a graph it looks like the teeth of a comb.
- **The space is small.** At low frequencies the cabin is smaller than the sound waves, which strongly boosts deep bass. This is called **cabin gain**. In the bass range, the cabin's dimensions also create peaks and dips.
- **Speaker locations are limited.** Speakers go where the car has room: low in the doors, in the dash, in the pillars beside the windshield, in the kick panels by your feet. Few of these are ideal.

- **It is noisy and it moves.** Road, tire and wind noise mask the music, especially the bass, and panels rattle and buzz.
- **Factory systems are complicated.** Many new cars process the sound before it reaches the speakers, with built-in EQ, delay and crossovers, which an upgrade has to work with or around.

That is why car audio leans so heavily on careful speaker placement, good installation and processing. The rest of this guide is about what Andy Wehmeyer has said on each of those, in his terms, with plain-English help along the way.

Start here: 50 things Andy says that almost nobody else does

Ten per topic, each with its key number or action and a reference to the full point and its sources.

In plain English: these are 50 of his points that go against, or well beyond, common car-audio advice. Each has his point in bold, a short note with his key number or action, and references (such as 1.1 #3) to the full points in the topic chapters, where the sources are listed.

Sound & placement (1 · Sound & speaker placement)

1. **Door speakers put left and right 180° out of phase near 250 Hz, in both front seats.** A ~28" path difference to the door mids causes it, right in the vocal range; people mistake those bands for "width". (1.1 #3, 1.9 #4)

   In plain English: Your left and right door speakers sit at different distances from your head (about 28 inches, roughly 70 cm, of difference), so around 250 Hz, a pitch in the middle of the human voice, the waves from the two sides arrive exactly opposite and cancel, in both front seats. He says the spread-out sound people praise as "width" is really these cancelling bands, which make you hear where the speakers are instead of a centered image.

2. **A car has no reverberant field, so you hear each speaker's power response: its output averaged over all angles.** Reflections arrive within ~20 ms and fuse with the direct sound, and no EQ or delay can undo a reflection's comb. (1.3 #1, 1.3 #4)

   In plain English: Every sound that bounces around a car reaches you within about 20 ms (twenty thousandths of a second) and blends with the direct sound, so what you hear is roughly the speaker's average output in all directions (its "power response"). Because the processor feeds the speaker before any bouncing happens, he says no EQ or delay setting can undo the dips a reflection causes.

3. **Predict reflection dips with a tape measure: a dip lands where the reflected path is half a wavelength longer.** A mid 6" from a surface dips near 500 Hz, so tuck mids into corners and keep tweeters away from surfaces. (1.3 #5)

   In plain English: A bounced copy of a sound travels farther than the direct sound; where that extra distance equals half a wavelength, the copy arrives upside down and cancels, leaving a dip. He says you can predict this with a tape measure (a mid 6 inches, about 150 mm, from a surface dips near 500 Hz), so push midranges into corners and keep tweeters away from surfaces, so the dip falls outside the range each speaker plays.

4. **A 6" + 1" two-way leaves an off-axis hole at its crossover, which is why cars need 3-ways.** Aiming only moves the hole; studio monitors get away with 2-ways because their walls are ~10 ft away. (1.4 #4, 1.4 #6)

   In plain English: A 6-inch woofer and a 1-inch tweeter spread their sound very differently near the crossover (the frequency where one hands over to the other), so off to the sides there is a "hole", and in a car that sideways sound reaches you by bouncing off surfaces right next to the speakers. He says aiming only moves the hole around, so cars need a 3-way (an added midrange); studio monitors get away with 2-ways because their room's walls are about 10 ft (3 m) away.

5. **Aim only the tweeter.** By his circumference ÷ wavelength rule, a 6" is omnidirectional below ~300–500 Hz and a 3" below ~3 kHz. (1.4 #2, 1.5 #1)

   In plain English: Low notes spread out evenly in all directions; higher notes start to "beam" forward. By his rule (compare the cone's circumference with the sound's wavelength) a 6-inch speaker spreads evenly below about 300–500 Hz and a 3-inch below about 3 kHz (3,000 Hz), so aiming those changes little in their working range, and only the tweeter, turned in a little, is worth aiming.

6. **Put a small midrange high in the door; an A-pillar is "a HORRIBLE place for a midrange".** A pod a foot off the pillar corner digs a hole near 250 Hz, and correct crossover phase gives height anyway. (1.6 #2, 1.6 #6, 1.7 #7)

   In plain English: He recommends a small midrange mounted high in the door and calls the A-pillar (the post beside the windshield) "a HORRIBLE place for a midrange": a pod hanging about a foot (30 cm) from the windshield, dash and side glass picks up reflections that dig a hole near 250 Hz. He says the height pillars promise comes anyway from high tweeters plus correctly phased crossovers, which pull the whole sound up to tweeter height.

7. **The stage can't be wider than your speakers.** Pan the balance fully left, then right: those are your real edges, and "width" from delay games is comb filtering. (1.7 #1, 1.7 #2)

   In plain English: The "stage" is the imaginary band of sound you hear spread across the dash. He says it can't be wider than the speakers themselves: pan the balance control fully left, then fully right, and where the sound lands are your true edges; extra "width" made by playing with delay is really comb filtering (a pattern of cancellations).

8. **Delay fixes one seat and doubles the error in the other; an all-pass near 250 Hz helps both.** His 2016 starting point: one 2nd-order all-pass near 250 Hz, Q about 1, on one side only. (1.9 #2, 1.9 #4, 1.9 #7)

   In plain English: The driver sits nearer the left speakers and the passenger nearer the right, so delaying one side to fix the driver's seat doubles the timing error at the passenger's seat. An all-pass filter shifts timing (phase) in one band without changing loudness, and near 250 Hz it helps both seats, not perfectly in either but better; his 2016 starting point was one such filter (2nd-order, Q about 1, a fairly broad setting) on one side only.

9. **"L+R is not a center."** A real center plays only what left and right share, and that content stays in L and R about 6 dB down. (1.10 #1, 1.10 #2)

   In plain English: "L+R" means simply adding left and right together into one center speaker, which he says is not a real center. A proper center plays only the sound that left and right have in common, and even then that shared sound stays in the left and right speakers as well, about 6 dB quieter.

10. **For rear fill, use plain in-phase stereo, then delay it until the stage returns to the front.** High-pass at 100 Hz or more, raise the rears until the stage drifts back, then add 10–20 ms (35 ms becomes an echo). (1.11 #4)

    In plain English: Rear fill means playing rear speakers quietly to add a sense of space without pulling the sound backwards. His recipe, after the front is tuned: feed the rears normal stereo with bass below at least 100 Hz filtered out, raise them until the stage starts drifting back, then add 10–20 ms of delay until it sits in front again; about 35 ms is long enough to be heard as an echo.

Installation & hardware (2 · Installation & hardware)

11. **A door midbass needs a sealed baffle, not a box; tiny boxes are "bass prevention".** With a high-pass, just isolate front from back: seal it to the trim panel or cover the sheet-metal holes with deadener. (2.1 #1, 2.1 #2)

    In plain English: A baffle is the panel a speaker is mounted on; a box encloses its back. He says a door midbass whose deepest bass is already blocked by a high-pass filter needs no box, only its front sound kept apart from the sound off its back (seal it to the door's trim panel, or cover the holes in the door metal with deadening sheets), and that tiny boxes behind it actually reduce bass ("bass prevention").

12. **Pull out foam cup "baffles", and stick foam rings to the bracket, never the speaker.** Foam stops rattles; it doesn't "make midbass", so keep at most a cup's top as a rain shield. (2.1 #6, 2.1 #7)

    In plain English: Foam cups are sold to go behind door speakers; he says pull them out, keeping at most the top part as a shield against water running down inside the door. Foam rings are only for stopping the door panel rattling, so stick them to the speaker's mounting bracket, never to the speaker, and don't expect them to add bass.

13. **Choose a flat, low-Q midbass, because the door and its corner already add the bass.** Each halving of radiating space adds ~6 dB below ~500 Hz, so a "full", high-Q driver turns into a peak. (2.1 #8, 2.1 #9)

    In plain English: "Q" describes how peaky a speaker's own bass response is; low-Q means flatter, with no built-in bass bump. He says the door and the corner it sits in already boost bass (about 6 dB below about 500 Hz each time the space the sound can spread into is halved), so a speaker designed to sound "full" ends up with an unwanted peak in a car.

14. **Midbass "punch" lives at 1–4 kHz, not in a bigger woofer.** Prove it on headphones: low-pass a punchy track at 100 Hz, then 200 Hz, and the punch is gone. (2.1 #13)

    In plain English: "Punch" is the snap you feel from kick drums and bass. His point is that it lives in the 1–4 kHz range (1,000–4,000 Hz: the thumb slap on a bass, the snap of toms), not in a bigger woofer; to prove it, play a punchy song on headphones with everything above 100 Hz, then above 200 Hz, filtered out, and the punch is gone.

15. **Leave room beyond Xmax: keep grilles about 1.25" off a woofer's front.** The cone travels past Xmax through the top-plate thickness, so a surround hitting a grille or panel is an install error. (2.1 #19)

    In plain English: Xmax is how far a speaker cone can move in and out while still behaving cleanly. He says the cone actually travels further than that, so keep grilles about 1.25 inches (about 32 mm) off a woofer's front; if its rubber edge hits a grille or panel, that's an installation mistake, not a faulty speaker.

16. **Deadening is rattle control: do the trim panel first, then the inner skin.** Play a 250 Hz tone: if nothing buzzes, a 250 Hz peak is a cabin reflection to cut with EQ. (2.2 #1, 2.2 #3)

    In plain English: Sound deadening means sticking heavy damping sheets and foam onto car panels; he treats it as rattle control, starting with the door's inner trim panel (the worst rattler) and then the inner metal skin. To check a loud spot at 250 Hz, play a steady 250 Hz tone: if nothing buzzes, the peak comes from the cabin's shape reflecting sound, and you turn that frequency down with EQ instead of adding deadener.

17. **Quieter tires were "the best midbass upgrade" he ever made.** For road noise, deaden first, then fit quieter tires, then save per-road DSP presets; never EQ for noise above 300 Hz. (2.2 #4)

    In plain English: He says swapping noisy tires for quieter ones was the best midbass improvement he ever made, because road noise drowns out bass. His order for road noise: deaden the car, then fit quieter tires, then save separate DSP settings (presets) for roads you drive often, and never use EQ to fight noise above 300 Hz.

18. **Map locations before cutting anything: a speaker at the driver's head, a mic at each candidate spot.** Sound paths work both ways, so this finds good sub, midbass and mid spots, but not tweeter spots. (2.3 #13)

    In plain English: Before cutting any holes, he "maps" the car: a small speaker goes where the driver's head would be, and a measuring microphone goes at each spot you might mount a speaker. Because sound paths work the same in both directions, this shows which spots suit subwoofers, midbass and midranges, though not tweeters, which spread their sound too narrowly for the trick to work.

19. **A pod is an enclosure, and a leaky one "rattles".** Size the pod to the driver, gasket it, and run an impedance sweep: a leak shows up as an extra, port-like resonance. (2.4 #1, 2.4 #3, 2.4 #4)

    In plain English: A pod is a custom housing for a speaker (on an A-pillar, for example), and he says it is really a small speaker box: it must be sized to suit the speaker, sealed with a gasket (a soft seal) and airtight, because a leaky pod buzzes. To check it, run an impedance sweep (a test of the speaker's electrical load across frequencies): a leak shows up as an extra bump like the one a ported box makes.

20. **Keep a cheap series capacitor on tweeters even in active systems, against amp failures and DSP typos.** Use about 68 µF for a 4 Ω tweeter as a "parking brake"; a 10 Ω resistor across it cures turn-on pops. (2.9 #1, 2.9 #3)

    In plain English: Even when a DSP and separate amps do the crossover (an "active" system), he keeps a cheap capacitor wired in line with each tweeter as a "parking brake" that blocks bass if an amp fails or someone types a wrong DSP setting, since bass quickly destroys a tweeter. His example is about 68 µF (microfarads) for a 4-ohm tweeter, and a 10-ohm resistor across the capacitor stops pops when the system turns on.

Tuning & measurement (3 · Tuning & measurement)

21. **Measure and fix everything first; listen once, at the end.** If hours of by-ear touch-up follow, the measurement, your reading of it or the process is wrong. (3.2 #1)

    In plain English: He tunes by measuring with a microphone and fixing everything the measurements show first, and listens once, at the end. If hours of tweaking by ear follow, he says something went wrong: the measurement, how you read it, or the method itself.

22. **One mic on the headrest, measured from outside the car with the door shut, is enough.** With 1/6-octave smoothing it gets about 90% of a spatial average; waving it around your head from the seat eats the treble. (3.4 #1, 3.4 #2, 3.4 #5)

    In plain English: He straps one measuring microphone to the headrest at the listening position, runs the cable outside and measures with the door shut. With 1/6-octave smoothing (averaging out tiny wiggles on the graph), that single mic gets about 90% of the accuracy of averaging many mic positions, whereas sitting in the seat waving the mic around your head lets your head block the treble and skews the graph.

23. **Leave narrow peaks and dips above ~1 kHz alone, and move the mic before fixing one.** If a feature vanishes when the mic moves a few inches, fixing it probably makes things worse. (3.4 #4, 3.4 #10)

    In plain English: Narrow peaks and dips above about 1 kHz come from sounds arriving at slightly different times, and he says we don't hear them because tiny head movements average them out, so leave them. Before "fixing" one, move the mic a few inches: if the feature vanishes, correcting it will probably make things worse.

24. **Put the target on the dips and cut the peaks down to it.** A 64 dB peak over a 57.5 dB dip gets a ~7 dB cut; then restore overall level with amp gain. (3.6 #3)

    In plain English: A "target curve" is the shape you want the frequency graph to follow. He says line the target up with the dips (the low points), not the peaks, and turn each peak down to meet it (with a dip at 57.5 dB, a 64 dB peak gets about a 7 dB cut), then raise the amplifier gain to get the overall loudness back.

25. **Matching left to right beats hitting the target.** Match within a couple of dB, and if one side has a dip you can't fix, copy it to the other side. (3.6 #2)

    In plain English: He says making the left and right sides measure alike, within a couple of dB, matters more than hitting the target exactly. If one side has a dip you can't fix, add the same dip to the other side, which protects the centered sound.

26. **Never significantly boost a null; first ask whether the dip responds to EQ.** Driver and box problems do; reflection dips don't, and a dip only with both sides playing is a phase or delay problem. (3.6 #9, 3.6 #10)

    In plain English: A "null" is a deep dip where sounds cancel; boosting it with EQ wastes amp power and strains the speaker, so he says never boost one by much. First ask whether the dip responds to EQ: dips from the speaker or its box do, dips from reflections don't, and a dip that appears only when both sides play is a timing (phase or delay) problem, not an EQ one.

27. **Target-curve wars are pointless: popular car curves differ by about 2 dB, and the "Harman curve" began as his.** His target today: about +10 dB below ~60 Hz, flat from ~160 Hz, and −6 dB at 20 kHz. (3.5 #1, 3.5 #2, 3.5 #3)

    In plain English: Hobbyists argue about which target curve is best; he says the popular car curves differ by only about 2 dB, and the so-called "Harman curve" began as his own installer curve, which he took to Harman. His target today: bass about 10 dB louder below about 60 Hz, flat from about 160 Hz up, easing down to 6 dB quieter at 20 kHz.

28. **Delay doesn't move speakers: center an off-center image with level, never delay.** Turn the whole side the image leans toward down a little; about 6 dB moves an image halfway. (3.9 #1, 3.10 #2, 3.10 #4)

    In plain English: Delay (holding back one speaker's sound by a few thousandths of a second) changes how speakers' sounds combine, but it can't change where the speakers seem to be. If a centered voice sits off to one side, he turns the whole side it leans toward down a little rather than using delay; about 6 dB of difference moves an image roughly halfway.

29. **Set delays with a tape measure, keyed to the midrange, then leave them alone.** Impulse peaks aren't arrival times for subs and midbass, and auto-alignment measures dash speakers via the windshield. (3.9 #3, 3.9 #5, 3.9 #6)

    In plain English: He sets delays by measuring with a tape measure from the headrest to each speaker, using the midrange's distance for the front speakers, and then leaves them alone. He says graph-based and automatic methods mislead: impulse peaks (spikes on a timing graph) don't show true arrival times for subs and midbass, and auto-alignment measures dash speakers by the sound bouncing off the windshield.

30. **Read phase from the left-plus-right sum of mono pink noise.** About +6 dB over one side is in phase, +3 dB ~90°, equal to one side ~120°; leave anything within ±90° alone. (3.8 #6, 3.8 #7)

    In plain English: Phase describes whether two speakers' waves move in step. Play the same pink noise (an even hiss containing all frequencies) through both sides and measure: about 6 dB louder than one side alone means in step, about 3 dB louder means roughly 90° apart, the same level as one side means about 120° apart, and he says anything within ±90° can be left alone.

Bass, power & factory systems (4 · Bass, power & factory systems)

31. **Cabin gain belongs to the car: build the most efficient box you can fit, then EQ.** Cut a boomy in-car peak by less than the graph says; he cuts 6–10 dB where it shows 15. (4.1 #1, 4.1 #6, 4.1 #8)

    In plain English: A closed car naturally boosts deep bass ("cabin gain"), and he says that boost belongs to the car, not to the subwoofer. So build the most efficient sub box that fits, then correct it with EQ; where the graph shows a boomy peak, cut by less than the graph suggests (he cuts 6–10 dB where it shows 15).

32. **Bass gives away its location through distortion and rattles, not arrival time.** Bass up front needs just two things: sub and midbass in phase at the crossover, and no rattles or distortion. (4.2 #1, 4.2 #2)

    In plain English: He says what gives away where bass is coming from is distortion and rattles, not when its sound arrives. For bass that seems to come from the front, you need just two things: the sub and midbass in phase where they hand over, and no rattles or distortion.

33. **Cross the sub at 60 Hz and door midbass at 80 Hz, both LR4; never run door midbass down to 40 Hz.** Lower the sub low-pass half an octave for every 12 dB you turn the sub up. (4.2 #6, 4.6 #1)

    In plain English: The crossover is where the sub hands over to the door midbass. His default is 60 Hz on the sub and 80 Hz on the midbass, both LR4 (steep, 24 dB-per-octave filters), because you will turn the sub up; never make door speakers play down to 40 Hz, and lower the sub's crossover by half an octave for every 12 dB you turn it up.

34. **He prefers vented sub boxes tuned for minimum excursion, with the in-car peak EQ'd out.** Short on space but not on power, go sealed plus EQ; with no EQ at all, sealed is probably better. (4.3 #1, 4.3 #4, 4.3 #6)

    In plain English: A vented (ported) box has an opening tuned to reinforce bass; a sealed box doesn't. He prefers vented boxes tuned so the cone moves as little as possible, with the boomy in-car peak removed by EQ; with little space but plenty of amp power he says go sealed plus EQ, and with no EQ at all sealed is probably better.

35. **Music averages about 1/32 of its peak power; give mids and highs 50–100 W and the sub at least ~800 W.** Only the sub amp ever reaches rated power; a tweeter crossed at 2.5 kHz gets ~0.16 W of a "20 W" calibration. (4.7 #1, 4.7 #2, 4.7 #3)

    In plain English: Music's average power is only about 1/32 of its loudest peaks, and most of it is bass, so his rule of thumb is 50–100 watts for mids and highs and at least about 800 W for the sub. Only the sub amp ever reaches its rated power: in his example, with the amp calibrated to 20 W at 20 Hz (deep bass), a tweeter crossed at 2.5 kHz gets about 0.16 W.

36. **Set the DSP input so it can never clip at full volume, and give only the sub amp a sine wave.** About 70% of GB25 warranty claims were DSP input clipping; a 4 V radio needs a 4–5 V input setting. (4.7 #6, 4.8 #1, 4.8 #4)

    In plain English: A DSP (digital sound processor) turns the incoming signal into numbers, and too strong an input "clips" (squares off the waves), which sounds like broken speakers and can damage them; he says about 70% of warranty claims on Audiofrog's GB25 speaker came from this. So set the DSP's input so it can't clip even at full volume (a radio putting out 4 volts needs a 4–5 V setting), and use a sine-wave test tone only for setting the sub amp.

37. **Amps with built-in DSP can't win back output lost to EQ cuts.** After tuning cuts, a "1,000 W" DSP amp may make ~10 W; for high-end systems use a separate DSP and amps with analog gain. (4.8 #9)

    In plain English: Some amplifiers have a DSP built in, and he says they cannot turn the signal back up after tuning has cut the EQ, so a "1,000 W" DSP amp may end up making only about 10 W. For high-end systems he recommends a separate DSP feeding amps that have their own analog gain controls.

38. **Hiss is not a ground loop; loops whine.** Ohm the RCA shield to the amp's power ground: near 0 Ω means single-ended (use coax), hundreds of ohms means differential (use twisted pair). (4.10 #1, 4.10 #5)

    In plain English: He says steady hiss is not a ground-loop problem; ground loops (faults in how equipment shares its ground connection) cause whine. To pick the right signal cable, measure the resistance from the RCA plug's outer shield to the amp's power ground: near 0 ohms means a "single-ended" input (use ordinary coax), hundreds of ohms means a "differential" input (use twisted-pair cable).

39. **Tell factory all-pass filters from factory delay before you add any delay.** On mono pink noise, one big L+R dip at 200–500 Hz means all-pass (add no delays); an even comb means delay. (4.14 #2, 4.14 #4)

    In plain English: Many factory systems already adjust timing, using either delay or all-pass filters (which shift timing in one band). He says find out which before adding any delay: play mono pink noise and look at left plus right together; one big dip between 200 and 500 Hz means all-pass filters (add no delay), and a row of evenly spaced dips (a "comb") means delay.

40. **In factory systems that upmix, keep the factory processing and wire each output one-to-one to new amps.** Then just re-EQ and set levels, and "DO NOT TRY TO SUM" a Logic 7-type car back together. (4.14 #11, 4.15 #4)

    In plain English: Some factory systems "upmix" stereo into many channels for different speakers (Harman's Logic 7 is one). He says keep that factory processing, connect each factory output one-to-one to a channel of a new amp, then just redo the EQ and levels, and in Logic 7-type cars "DO NOT TRY TO SUM" (recombine) the channels back into plain stereo.

Myths & buying (5 · Myths, buying & industry)

41. **"You cannot blow a speaker with too little power."** Clipping pushes a wave toward a square, roughly doubling its average power, and all the extra is treble that cooks tweeters. (5.1 #26)

    In plain English: Many people believe an underpowered amp blows speakers; he says it cannot. The damage comes from clipping (driving an amp, radio or DSP past its limit), which squares off the waves, roughly doubles their average power and puts all the extra into treble that overheats tweeters.

42. **A correct tune matters more than $200 versus $2,000 speakers.** His priorities are processing, then installation, then brand; EQ is the only thing that overcomes the car. (5.3 #2)

    In plain English: He says setting up (tuning) a system correctly matters more than whether the speakers cost $200 or $2,000. His order of priority is processing (DSP and EQ), then installation, then brand, because only EQ can overcome what the car does to the sound.

43. **Upgrade a factory system in stages: speakers on the factory amp and EQ first, then power, then sub, rears and DSP.** Only about 10–20% of a factory EQ corrects the stock speakers, so with similar new speakers it keeps doing useful work. (5.3 #6, 4.16 #1)

    In plain English: For a car with a factory system, he suggests upgrading in steps: new speakers on the factory amp and its built-in EQ first, then more amplifier power, then a sub, rear speakers and a DSP. Only about 10–20% of a factory EQ is there to correct the original speakers, so with similar new speakers the rest of it keeps doing useful work.

44. **Go to the best shop in town and ask it for a mid-level system.** Paying ~30% more for the shop that can "build the space shuttle" beats pushing a cheap shop past its level. (5.4 #1)

    In plain English: He advises going to the best shop in town and asking it for a mid-level system, telling them your budget, rather than pushing a cheap shop to work beyond its skill. In his view paying about 30% more for the shop that could "build the space shuttle" is worth it.

45. **"Correct" is the beginning of expertise; backing off it for the customer is the rest.** His disco fan boosted bass ~12 dB at the head unit and kept blowing speakers; a higher, pro-style crossover fixed it. (5.4 #15)

    In plain English: His point is that knowing the technically correct setup is only the start; expertise is also knowing when to back off it to suit the customer. His example: a 1970s-disco fan kept boosting bass about 12 dB at the radio and blowing speakers, and the fix was a higher crossover like those in pro sound systems, giving up up-front bass the customer didn't care about.

46. **Learn what imaging is with a matched $5 thrift-store pair of speakers.** Sit centered in an equilateral triangle, then flip one's polarity; that, not a concert, is a tuner's first reference. (5.7 #5)

    In plain English: "Imaging" is hearing sounds appear at exact spots between the speakers. He says learn it with any matched pair of cheap speakers (about $5 at a thrift store): sit centered so you and the two speakers form an equal-sided triangle, then swap the wires on one speaker (flip its polarity) and hear the difference; to him, that, not a live concert, is the first reference a tuner needs.

47. **Distortion on only some tracks is usually in the source.** Many streaming masters are clipped and phone "volume levelers" add more; if it stays with the phone turned well down, it's the track or codec. (5.7 #12)

    In plain English: If only some songs sound distorted, he says the problem is usually in the recording: many streaming masters are already clipped, and phone "volume leveler" settings add more. If the distortion stays even with the phone's volume turned well down, it's in the track or the codec (the audio compression format).

48. **Damping factor above ~10 and channel separation above ~60 dB buy nothing in a car.** A damping factor of 1000 moves Qts only from 0.707 to 0.708, and each ear already hears the opposite speaker about 6 dB down. (5.1 #19, 5.1 #24)

    In plain English: Damping factor (how firmly an amp controls a speaker) and channel separation (how little of one channel leaks into the other) are often advertised. He says beyond about 10 and about 60 dB they make no difference in a car: a damping factor of 1000 changes a speaker's Qts (a measure of its bass behavior) only from 0.707 to 0.708, and each ear already hears the opposite speaker only about 6 dB quieter.

49. **Break-in matters mainly for subwoofers.** It's the spider's resin cracking, which matters at and below resonance; mids and tweeters crossed above it barely change. (5.1 #30)

    In plain English: "Break-in" is the idea that new speakers change as they are played. He says the real change is resin in the spider (the corrugated fabric that keeps the cone centered) cracking and loosening, which matters at and below the speaker's resonance (its natural bass frequency), so subwoofers change but midranges and tweeters crossed above that barely do.

50. **Know when to stop: "imagine how much better..." is the biggest threat to satisfaction.** Log tracks that annoy you, check that band with pink noise and fix only real peaks, or you end with a "smile" curve. (5.7 #13)

    In plain English: He warns that "imagine how much better..." is the biggest threat to being happy with your system. Instead, note songs that annoy you, check that frequency band with pink noise and fix only real peaks; otherwise you keep cutting until you end with a "smile" curve (bass and treble up, middle scooped out) that sounds bad.

Where Andy changed his mind, and where sources conflict

In plain English: over the years he changed his advice on some topics, and a few of his sources say different things. The table shows each topic's earlier and later view with years, and the list after it shows where sources disagree; neither is ranked here, so read the full points under "See" for context.

His advice shifted on 26 topics over the years; the table lists them newest change first, and source disagreements follow it.

Topic	Earlier view (year)	Later view (year)	See
What he teaches	Tuning (about 20 years, before 2023)	System design: map the car with a mic and speaker before choosing locations (2023)	3.2 #15, 5.8
Rear fill	Differential (L−R) fill, delayed 12–20 ms (2017)	Plain in-phase stereo, high-passed and delayed 10–20 ms; L−R sounds "phasy" (2021)	1.11 #5, 3.9 #18
REW settings	FFT 16384, rectangular window, "dB not dBFS" (Tuning PDF, ~2018)	32k FFT; window type and dB vs dBFS make no difference (2021)	3.3 #5
Speaker break-in	Not needed (2015)	Real, but it is the spider (2017); matters only for subs (2021)	2.8 #6
Efficiency vs power	"Efficiency is overrated. Power is cheap." (2008–13, arguing for active crossovers)	Efficiency first in very loud systems (2021); a change of context more than a reversal	4.7 #11, 5.2 #10
All-pass on sub and midbass	Never made one work (2010–13); used them in his own car (2015–16)	Fix the crossover instead; if you must, put it on the high-pass side (2021)	4.2 #15
Dash-top center	A response "nightmare" that still raised the stage (2010), yet already his preferred center in the MS-8 FAQ (2011); a refinement, not a reversal	Mount it close to the windshield–dash junction; its ~2 kHz dip helps (2021)	2.3 #11
Midbass Q (partial)	Car midbass should be low-Q (2015; Jun 2021 blog)	Crossed above resonance, its Q "is not really important" (Sep 2021)	4.6 #3
Two-seat imaging	Phase EQ inaudible next to delay (2013–15); passenger seat a trade-off while delay was the only tool	Drop L/R delay for phase EQ (2016); it needs fiddling, so he won't teach it (2021); "forget the passenger" is a cop-out (2023)	1.9 #4, 1.9 #10, 3.8 #15, 3.9 #14
Factory integration	Sum, de-EQ and remove factory delay with two processors (2013–16)	Keep the factory all-pass filters and processing; wire outputs one-to-one (2021–23)	4.14 #11
Gain setting	Scope and −10 dB sine, with ~9–10 dB of deliberate clipping overlap (2008–15); by ear (2014–16)	Sine on the sub amp only, the rest by pink noise and RTA; overlap "never appropriate" at the DSP input (2018–22)	4.8 #4, 4.8 #5, 5.1 #27
Uncorrelated pink noise	Suggested for 31-band adjustments (2014)	Useless except for upmixers, L−R rears or gaming an RTA (2020–22)	3.3 #4
8" door woofers	Audiofrog would make no 8" and offered a 6×9 (2016)	The shallow GS8ND2 (2020), framed as a woofer crossed near 300 Hz	2.1 #15
Stage height	Comes from physical location (2016)	Crossover phase also holds everything at tweeter height (2020–21)	1.7 #7

Topic	Earlier view (year)	Later view (year)	See
Sub/midbass crossover	Both near 80 Hz LR4 (2010–15, MS-8 era)	Sub 60 Hz, midbass 80 Hz, because the sub gets boosted (2018–22)	3.7 #9, 4.2 #6
Sub delay	"Don't time-align subs" (2009–10); delay everything to the sub (2015)	Tape-measure the sub like any other speaker (2018–23 PDFs)	4.2 #12
Tweeterless widebanders	Must be on-axis (2012–14)	Aim them at the windshield; on-axis sounds "clinical" (2018–21)	1.8 #4
Main measuring tool	REW sweeps work "much better" (2012)	A live RTA on mono pink noise for production tuning (2018 on)	3.3 #1
Target bass lift	+6 to +9 dB (2008); +9 dB (MS-8, 2011)	+10 dB (2018 PDF); the curve's shape never changed	3.5 #4
Tuning order	EQ before delay (2011)	Delay first (2013 on); with one EQ per driver, EQ each driver to its crossover target first (2018)	3.1 evolution #2–#6
Single mic vs average	One mic is worse than an MS-8-style average (2013)	A smoothed single mic is ~90% as good, fine for production (2017–21)	3.4 #5
Matrix upmixers for music	"The ticket" for 2-channel music (2008–10)	Poor for music; use upmixers that extract common content (2017 on)	1.12 #6
How much speakers matter	Little, unless badly designed (2008–11)	No DSP removes a bad speaker's distortion (2016 on)	5.3 #3
Windshield corner	Keep speakers, mids especially, away from the junction (2012, still in Mar 2013)	Tuck mids tight into the corner; a pod hanging near it is the problem (Aug 2013, firmly 2021)	1.6 #8
Firing mids at the glass	Did it in his own car (2009)	Narrows the stage (2011); "the most dense comb filter possible" (2021)	1.6 #23
Tweeter aiming	Co-invented an aimable swivel tweeter at Infinity (filed 1997)	Sail panels; aim only the tweeter, and only a little (2011–21)	2.5 #7

Where his sources disagree:

- **Audiofrog's founding year:** sources give 2011, 2012 or 2014. Products shipped by mid-2015, and a 10th anniversary was marked in May 2023 (5.8).
- **Tweeter protection cap:** Tuning PDF v1 puts its −3 dB point about an octave below the tweeter's Fs; the amp manual says 50 µF plus a resistor. PDF v2 (2023) drops the step, though his Dec 2020 blog recommends a cap (2.9 #2, 3.1 #1, 3.7 #18).
- **Tweeter crossover:** documents give 1.5–2× Fs, about 2× Fs, 2.2 kHz, 2.5 kHz or 3–3.5 kHz, all well above Fs (2.9 #7, 3.7 #2, 3.7 #7). His claim that LR4 protects better than even 48 dB/oct rests on power near the filter's knee. The research notes could not square it with attenuation an octave lower (2.9 #8, 5.1 #57).
- **Sub delay:** some 2013–21 posts and a 2020 blog say not to time-align the fronts to the sub. The Tuning PDFs and other 2021 posts tape-measure the sub like any speaker (3.9 #12, 4.2 #12).
- **Tuning PDFs vs his other material:** the PDFs set the midrange at 0 dB and show a Q 10 cut at 1.6 kHz. His 2021 video and posts put the target on the dips, and his blog avoids Q above ~6 (3.1 #7, 3.6 #3, 3.6 #7). The video and PDFs also number the test-CD noise tracks differently (3.11 #3).
- **The attributed DIYMA gain guide vs his own posts:** the guide starts DSP channels at −12 dB and uses "7 drum beats" for centering. His 2022 post starts at −6 dB, and he says drum tracks can't check midbass (3.1 #2, 4.8 #4, 3.11 #4).

- **Mids aimed at the glass:** in Sep 2021 he condemned firing mids up at the windshield. From Oct 2021 he praised windshield-aimed factory dash "middlers", a factory-location strategy (1.6 #9, 1.6 #23, 2.3 #9).
- **A-pillars and crossfiring:** he calls pillars "a HORRIBLE place for a midrange", yet Audiofrog copy sells "the ultimate active system for A-pillars" (1.6 #6). He calls crossfired pillar tweeters a "band-aid"; a 2015 Harman/JBL/Infinity tip sheet of unconfirmed authorship recommends it (1.5 #6).
- **Where bass "impact" comes from:** the midrange near 3 kHz in his 2009 and 2021 posts, but correct midbass-to-sub phase on an Oct 2021 dealer livestream (4.2 #3).
- **Smaller discrepancies:**
  - CES 1996 class-D blind demo: three listeners after 700–800 demos (told in 2022) vs four after ~1,500 (2023) (4.9 #3).
  - Harman patent US6876748: he recalls 250 Hz; its text cites about 200 Hz (1.9 #5).
  - MS-8 auto-EQ: 8 biquad filters per channel (2011) vs an FIR-like correction convolved with the audio (2017, 2022) (3.12 #2).
  - 1st-order all-pass: flips polarity below its corner (2015) vs pulse checkers show no reversal (2021) (4.14 #3).
  - Xmax: one-way overhang "with no +10% fudge" at JBL (2008) vs (coil − gap)/2 + 15% at Audiofrog (2020) (4.3 #21).
  - Lexus tap point: a secondhand ClubLexus report has MS-8 working only when tapped before the factory amp, against his usual advice (4.13 #3).

SECTION 1

1. How sound works in a car, and where to put the speakers

Andy Wehmeyer's placement advice starts from how stereo and hearing work, then follows the physics of a small, reflective cabin: which reflections matter, how each driver's size sets its dispersion, and where that puts each speaker. It ends with the special cases he cares most about: two good seats, center channels, rear speakers, upmixing, and "believability" over accuracy.

What this section is about: This section covers the basics behind every placement decision. It explains how two speakers create the illusion of sound coming from places where there is no speaker, and how our ears and brain work out where a sound comes from. It then covers what the hard, close surfaces of a car do to the sound and how a speaker's size decides how widely it spreads sound, and therefore where it should go. Later parts of the section deal with special cases such as two good seats, center channels and rear speakers.

1.1 How stereo imaging works, and why a car images for one seat

What this part is about: How two speakers can make a singer seem to stand in the empty space between them, and why that trick works properly only for a listener sitting exactly in the middle, which nobody in a car does. It also covers what delay (time alignment) can and cannot fix.

1.1 #1 A stereo image needs left and right to reach you matched in level, frequency response and phase.

> Stereo was built to place sound between two speakers for a centered listener; in a car nobody sits centered, so the near speaker arrives first and louder and pulls the image. Match the three conditions by measurement first; listen last.
>
> **Sources:** Audiofrog blog Dec 2020 (Time Alignment Part 1) · CMA Connected Oct 2021 @39:51 · DIYMA 2017-06-28

In plain English: A "stereo image" is the illusion that sounds come from places between and around two speakers, such as a singer who seems to stand in the middle of the dashboard where no speaker is. Three terms matter here. Level means loudness. Frequency response means how loud each pitch is, from deep bass to high treble. Phase means whether the ups and downs of two sound waves arrive in step. His point is that the illusion works only when the left and right sounds reach you equally loud, with the same tonal balance, and in step. Stereo was designed for a listener sitting centered between two speakers. In a car you sit off to one side, so the nearer speaker's sound arrives sooner and louder and pulls the image toward it. So in practice: match level, response and phase by measuring first, and use your ears last.

1.1 #2 Level places sounds between the speakers: about 6 dB moves an image halfway to one side.

> Left-only content stays in the left speaker and content common to both forms the phantom center. Time alignment fixes only the phase part, so a 20 dB left/right response mismatch (one 2017 case, at 80 Hz) still won't image.
>
> **Sources:** EDUCAR · DIYMA 2017-09-12

In plain English: When a recording is mixed, each instrument is placed left or right by making it louder in one channel than the other. A dB (decibel) is the unit for a difference in loudness. He says a difference of about 6 dB moves the image about halfway from the center toward one speaker. Sound that is only in the left channel stays at the left speaker. Sound that is equally in both channels forms a "phantom center", an image in the middle where no speaker exists. Time alignment means delaying the nearer speakers so both sides arrive together, and he says it fixes only the timing (phase) part. It cannot fix a big loudness mismatch: in one 2017 case, left and right differed by 20 dB at 80 Hz (a low bass note), and that would not image no matter how the delays were set.

1.1 #3 Door speakers put left and right 180° out of phase near 250 Hz, right in the vocal range.

> A ~28" path difference to door mids puts the first null at ~250 Hz and its odd multiples (750 Hz, 1.25 kHz...), where you hear speaker locations, not a center. A 1 ms error does the same from 500 Hz; those out-of-phase bands are what people mistake for "width". RARE
>
> **Sources:** Audiofrog blog Dec 2020 (Time Alignment Part 2) · DIYMA 2015-09-10 · DIYMA 2017-10-01

In plain English: "180° out of phase" means that when one speaker's wave pushes, the other's pulls, so at your ears they work against each other. In his example, the far door speaker is about 28 inches (about 70 cm) farther from you than the near one. At about 250 Hz, a pitch in the lower part of the human voice, that extra distance makes the two sides arrive exactly opposite. It happens again at 750 Hz, 1.25 kHz and so on. In those bands you hear the two speakers in their own places rather than one image in the center. He adds that a timing error of 1 ms (one thousandth of a second) does the same thing starting at 500 Hz. Many people like a "wide" sound and think it is good. He says those out-of-phase bands are exactly what people mistake for "width".

1.1 #4 A phantom image is vector addition, and arrival time "rotates" it toward the near speaker.

> That's why a balance control helps only at frequencies already in phase. Align both sides to one point between the ears: even two speakers both on your right image between them, because long wavelengths wrap around the head. RARE
>
> **Sources:** DIYMA 2015-10-19 · DIYMA 2015-10-19 (one side) · DIYMA 2015-10-19 (wrap)

In plain English: A vector is a quantity with both a size and a direction, like an arrow. His point is that the image you hear between two speakers comes from adding the left and right sounds together like arrows, so both loudness and timing count. A difference in arrival time "rotates" the result toward the nearer speaker, so the image swings that way. That is why the balance control, which only changes loudness, helps only at frequencies where the two sides already arrive in step. His fix is to set delays so both sides line up at one point between your ears. He adds that this works even if both speakers are on your right: you still hear an image between them, because low sounds have long waves that bend around your head and reach both ears.

1.1 #5 Phase decides whether two speakers sound like one source or two.

> Out of phase you hear both locations; in phase you hear one: midway for a left/right pair, at the higher-frequency driver for a crossover pair. That is how bass gets pulled forward to the tweeters, and why group delay is "less a curse than people think". RARE
>
> **Sources:** DIYMA 2021-12-02 · DIYMA 2021-09-07

In plain English: When two speakers play the same sound and it reaches you in phase (in step), your brain hears one source. Out of phase, you hear two sources in two places. For a left/right pair, the single source appears midway between them. A crossover is the filter that splits the music by pitch, sending low notes to a bigger speaker and high notes to a smaller one such as a tweeter. For a crossover pair that is in phase, he says the single source appears at the higher-frequency (smaller) driver. That is how bass from a low-mounted speaker can be "pulled forward" to where the tweeters are. Group delay is the small extra delay that crossovers and speakers give some frequencies, and many hobbyists worry about it. Because a well-designed crossover pair is heard at the upper driver anyway, he calls it "less a curse than people think".

1.1 #6 Delay fixes one seat: delay the near side (about 2 ms for 28"), and the center lands midway between the speakers, not straight ahead of you.

> One sample at 48 kHz is ~0.29", so quarter-inch steps are ample. Set delays to the midrange's distance, not the tweeter's or an average (phase matters most at 200–300 Hz; averaging "makes them BOTH wrong"). RARE
>
> **Sources:** Time Alignment Part 1 · DIYMA 2017-07-07 · DIYMA 2021-09-26

In plain English: A DSP (digital sound processor) can hold back the signal to a speaker by a tiny amount of time. You delay the nearer speaker so its sound arrives together with the far one. He says about 2 ms (two thousandths of a second) matches about 28 inches (about 70 cm) of extra distance. This only fixes one seat, and the center image then lands midway between the two speakers, not straight ahead of your face. A DSP works in tiny steps called samples. At 48 kHz (48,000 samples per second), one sample equals about 0.29 inch (about 7 mm) of distance, so quarter-inch steps are fine enough. He says to set the delays using the midrange speaker's distance, not the tweeter's and not an average of the two. His reason is that phase matters most around 200–300 Hz, and averaging "makes them BOTH wrong".

1.1 #7 Delay does not move speakers.

> The DSP-delay diagram used for ~30 years is "seriously misleading": delay changes only what happens between the speakers, never where a speaker or the stage edge appears. Test: play only the left speaker, change its delay with eyes closed, and nothing moves. RARE
>
> **Sources:** Time Alignment Part 1 · Time Alignment Part 2

In plain English: For about 30 years, a common diagram has explained time alignment by showing the delay "moving" the speakers so they all sit the same distance from you. He calls that diagram "seriously misleading". He says delay changes only what happens between the speakers, meaning where the phantom images land. It never changes where a speaker itself seems to be, or where the edge of the stage is. His test: play only the left speaker, close your eyes, and change its delay. Nothing moves. (With one speaker playing, there is no second sound for the delay to be early or late against.)

1.1 #8 Image off-center after tape-measure delays? Fix it with level, never by re-tweaking delay.

> Delay tweaks only add comb filtering. As the last step, turn one whole side up or down until the singer is centered, ignoring RTA numbers (the mic can't separate direct sound from reflections); for a left-hand-drive driver the right side usually ends up louder. It has worked in "hundreds of cars". RARE
>
> **Sources:** DIYMA 2017-07-07 · DIYMA 2018-05-22 · DIYMA 2022-03-19

In plain English: "Tape-measure delays" means measuring each speaker's distance to your head with a tape measure and setting the delays from those numbers. If the singer still isn't centered after that, he says not to keep fiddling with the delays. Each tweak only adds comb filtering, a row of dips in the sound caused when two copies of it arrive slightly apart. Instead, as the very last step, turn one whole side up or down until the singer is centered. An RTA (real-time analyzer) is a microphone and display that shows loudness at each frequency. Ignore its numbers here, he says, because the microphone can't tell the direct sound from the reflections. For someone driving a left-hand-drive car, the right side usually ends up louder. He says this has worked in "hundreds of cars".

1.1 #9 Delays chosen to make drivers sum at a crossover also shift the left-right image.

> Left-right position is set mainly by level and phase at the ears; for one-seat imaging use tape-measure delays, LR4 (Linkwitz-Riley 24 dB/oct) slopes and EQ, or an all-pass filter (a filter that shifts phase without changing level). Exception: with a real center and upmixer, crossover-optimizing delays are fine. RARE
>
> **Sources:** DIYMA 2015-10-19 · DIYMA 2015-07-24

In plain English: Some tuners set delays not from distance but to make two drivers on the same side, such as a woofer and a tweeter, blend smoothly ("sum") at their crossover point. He says those delays also shift where sounds land between left and right, because left-right position is set mainly by level and phase at your ears. For imaging in one seat, his recipe is tape-measure delays, LR4 crossover slopes and EQ, or an all-pass filter. LR4 (Linkwitz-Riley, 24 dB per octave) is a standard crossover filter shape. EQ (equalization) means turning particular frequencies up or down. An all-pass filter shifts phase without changing loudness. The exception is a car with a real center speaker and an upmixer, a processor that creates a center signal from a two-channel recording. There, he says, delays chosen to optimize the crossover are fine.

1.1 #10 A correct center sounds "too loud" or "mono" at first, because we've learned to like fake width.

> Small path errors keep fundamentals centered but throw harmonics to the speakers, like a boombox "stereo wide" switch. Of two famous producers he demoed a strong-center car to, one loved it and one called it "mono"; listeners usually come round to the stable center. RARE
>
> **Sources:** DIYMA MS-8 FAQ #10 · CMA Connected Oct 2021 @41:19 · DIYMA 2013-01-24

In plain English: "Mono" means a single channel, with everything coming from one place. He says a properly centered image often sounds "too loud" in the middle, or "mono", the first time you hear it, because we have learned to like fake width. A musical note has a fundamental (its main pitch) and harmonics (higher overtones). Small errors in path length keep the low fundamentals centered but throw the higher harmonics out toward the speakers. He compares this to the "stereo wide"

switch on a boombox. He demonstrated a car with a strong center to two famous record producers: one loved it, and the other called it "mono". In his experience, listeners usually come round to preferring the stable center.

1.1 #11 Ultra-precise high-frequency alignment is right at only one point in space.

> Move the mic an inch and drivers are out of phase again; 1 mm delay resolution is "nonsense" and laser-aligned acoustic centers are "garbage" where heads move. Fussing over mid-to-tweeter crossover phase is "ridiculous"; sub/midbass and midbass/mid just need to be far from 180°. RARE
>
> **Sources:** OFCAP Ep. 19 @2:56:00 · DIYMA 2015-05-29 · DIYMA 2021-07-12

In plain English: High-pitched sounds have very short waves, about an inch or less at the top of the treble range. So timing that is perfect at one exact spot is wrong a little way off. He says that if you move the microphone an inch, the drivers are out of phase again. That is why he calls 1 mm delay resolution "nonsense". He calls aligning speakers' acoustic centers with a laser "garbage" in a car, where heads move. He calls fussing over the phase at the midrange-to-tweeter crossover "ridiculous". For the lower crossovers, subwoofer-to-midbass and midbass-to-midrange, he says the two drivers just need to be far from 180° (fully opposite), not perfectly matched.

1.1 #12 The goal is one apparent source above the dash, not a "rainbow" of bands in different places, and the cure is a good crossover.

> What counts is arrival at your ears, not at the baffle (a mid 6.8" farther than the tweeter is 180° out at 1 kHz). Right phase at sub–midbass puts bass in front; right phase at midbass–mid–tweeter holds vocals at tweeter height. RARE
>
> **Sources:** Audiofrog blog Dec 2020 (Time Alignment Part Three) · DIYMA 2021-11-03 · DIYMA 2015-05-29

In plain English: He wants all the music to seem to come from one source, up above the dashboard. A "rainbow" is when different pitch bands seem to come from different places: bass low in the doors, voices somewhere else, cymbals at the tweeters. His cure is a good crossover. A baffle is the panel a speaker is mounted in. He says what counts is when each driver's sound arrives at your ears, not where the speakers sit on the baffle. For example, a midrange 6.8 inches (about 17 cm) farther away than the tweeter arrives 180° out of phase (exactly opposite) at 1 kHz. When the subwoofer-to-midbass crossover is in the right phase, the bass seems to come from in front. When midbass, midrange and tweeter are in the right phase, voices stay up at tweeter height.

1.1 #13 In a car, left plus right sums to +6 dB only in the midbass; higher up, about +3 dB is normal.

> The dense reflection comb averages to ~90° between sides, so leave anything within ±90° alone (true cancellation is 135–225°). Every peak or dip is a phase shift, so mismatched peaks mean mismatched phase. If the center is solid and pans move smoothly, ignore a small dip. RARE
>
> **Sources:** DIYMA 2021-08-26 · DIYMA 2021-08-27 · DIYMA 2022-03-24

In plain English: Two equal sounds that arrive exactly in step add up to 6 dB louder than one alone. If they are 90° apart, they add up to about 3 dB louder. He says that in a car, left plus right reaches +6 dB only in the midbass. Higher up, about +3 dB is normal. The reason is the dense pattern of reflections (the "comb"), which averages out to about 90° of difference between the two sides. So if the two sides are within ±90° of each other, leave them alone. Real cancellation happens only at 135–225°, around fully opposite. He says every peak or dip in a response is a phase

shift, so peaks that don't match between left and right mean the phase doesn't match either. So in practice: if the center image is solid and sounds move smoothly across when panned, ignore a small dip.

1.1 #14 Learn what imaging is with a $5 pair of identical speakers.

> They needn't be good, only the same: set a thrift-store pair in an equilateral triangle, sit in the middle, then flip one's polarity to hear out-of-phase. That, not "knowing live music", is a tuner's first reference; a car needs extra steps only because you can't sit in the middle. RARE
>
> **Sources:** Absolute Electronix 2023 @41:56 · Absolute Electronix 2023 @46:15

In plain English: He suggests a cheap experiment to learn what a stereo image is. Buy any pair of identical speakers, for example from a thrift store for about $5. They don't need to be good, only the same. Set them up with your listening spot so that the two speakers and your head form an equilateral triangle (all three sides the same length), and sit centered. Listen to how voices appear in the middle. Then flip the polarity of one speaker (swap its + and − wires) to hear what out-of-phase sounds like. He says this, not "knowing live music", is a tuner's first reference. A car needs extra tuning steps only because you can't sit in the middle.

1.2 How we hear direction (the psychoacoustics he builds on)

What this part is about: How our ears and brain work out where a sound comes from, and which of those methods work at which pitches. He builds his placement and tuning advice on these facts, so this part explains why some errors matter a lot and others hardly at all.

1.2 #1 His localization map by band.

> Below ~60–80 Hz neither time nor level cues work; ~80–800 Hz we use arrival-time (phase) differences between the ears; ~800 Hz–1.6 kHz is a weak zone ("we just basically suck at it"); above that, level only. So arrival matters most for midbass and mids; phase hearing fades from ~1 kHz and is gone by ~6 kHz. RARE
>
> **Sources:** DIYMA 2013-11-20 · DIYMA 2016-04-27 · DIYMA 2016-03-17

In plain English: Localization means working out where a sound is coming from. He splits the range of pitches into zones. Below about 60–80 Hz (deep bass), neither timing nor loudness differences between your two ears help, so you can't tell where the sound is. From about 80 Hz to 800 Hz, we use differences in arrival time (phase) between the two ears. From about 800 Hz to 1.6 kHz is a weak zone where, in his words, "we just basically suck at it". Above that, we use only level (loudness) differences between the ears. So arrival timing matters most for the midbass and midrange. Our hearing of phase fades from about 1 kHz and is gone by about 6 kHz.

1.2 #2 Only the lowest comb-filter notch really matters; head movement smears the rest.

> A comb filter is the row of dips two delayed copies of a sound create. The first notch is widest; higher ones are narrow, and narrow dips are far harder to hear than peaks. So judge with band-limited pink noise or music, never sine waves. RARE
>
> **Sources:** DIYMA 2013-11-20 · DIYMA 2017-07-12 · DIYMA MS-8 FAQ #21

In plain English: The point defines a comb filter. It is called that because, on a graph, the row of dips looks like the teeth of a comb. He says the first, lowest-pitched dip is the widest, and only that one really matters. The higher dips are narrow, and as your head moves they shift and smear out. Narrow dips are also far harder to hear than peaks. So in practice: judge the sound by listening to band-limited pink noise or to music, never to sine waves. Pink noise is a steady rushing sound containing all pitches; "band-limited" means it is cut down to one range of pitches. A sine wave is a single pure tone.

1.2 #3 Above ~1 kHz the in-car response swings 25 dB inside a head-sized circle, and you don't hear it.

> Five mic positions in a 7" circle agree below ~200 Hz, diverge from 200 Hz to 1 kHz and swing ≥25 dB above it. The success of kick-panel cars suggests path errors above ~1 kHz matter much less. RARE
>
> **Sources:** Audiofrog blog Jan 2021 (Measuring Frequency Response) · Time Alignment Part 2

In plain English: He measured with a microphone at five spots in a circle 7 inches (about 18 cm) across, about the size of a head. Below about 200 Hz the five measurements agreed. From 200 Hz to 1 kHz they started to differ. Above 1 kHz they swung by 25 dB or more, a huge difference, between spots only inches apart. Yet we don't hear the sound changing like that as we move our heads. He also points to cars with speakers in the kick panels (the panels low down in front of the doors, by your feet). Kick-panel mounting makes the left and right distances more nearly equal, which pushes the remaining errors up to higher pitches. He takes the success of such cars as a sign that path-length errors above about 1 kHz matter much less.

1.2 #4 Precedence: sounds arriving within ~30 ms fuse and are heard at the first arrival.

> That's why a rear sub can image in front (boost it enough and you hear it behind you), and why every in-car reflection, all within ~20 ms, fuses with the direct sound. RARE
>
> **Sources:** DIYMA 2014-03-22 · DIYMA 2009-12-06

In plain English: The precedence effect (also called the Haas effect) is how the brain handles echoes. When the same sound reaches you more than once within a short time, you hear a single sound, placed where the first arrival came from. He gives that window as about 30 ms (thirty thousandths of a second). He says this is why a subwoofer in the back of the car can seem to play from the front. But boost it enough and you will hear it behind you. It is also why every reflection inside a car, all of which arrive within about 20 ms, blends with the direct sound instead of sounding like an echo.

1.2 #5 "HRTF EQ" can't raise the stage.

> HRTF (head-related transfer function) is the head-and-ear filtering that cues direction; it works only with a moving head or source. EQ a "height" bump (e.g. 3.5 kHz) and the image rises only while you're adjusting, then settles back, leaving a bump. Height comes from where speakers are (1.7 #7).
>
> RARE Era: unchanged 2008–2021
>
> **Sources:** DIYMA 2008-07-27 · DIYMA 2016-04-27 · DIYMA 2021-11-11

In plain English: The point defines HRTF. The folds of your outer ear change the tone of a sound slightly depending on its direction, and that is one way we sense height. Some tuners try to fake a higher stage with EQ, for example by adding a bump around 3.5 kHz. He says the image rises only

while you are adjusting the EQ. Then it settles back down, and all you are left with is a bump in the response. These cues, he says, only work when your head or the sound source is moving. Real height comes from where the speakers actually are (see point 1.7 #7). The tag notes that he held this view unchanged from 2008 to 2021.

1.2 #6 We hear height worst, yet car audio prioritizes stage height.

> We localize front sounds well, rear sounds poorly and height worst. Hanging mids in A-pillars optimizes what we hear worst while distorting what we hear best, like painting a tonally perfect picture for a color-blind viewer with every shape wrong. RARE
>
> **Sources:** DIYMA 2016-04-27 · DIYMA 2013-11-20

In plain English: He says we locate sounds in front of us well, sounds behind us poorly, and height worst of all. Many people believe the most important thing is a high soundstage, and they mount midrange speakers in the A-pillars, the posts on either side of the windshield. He says this optimizes the thing we hear worst while distorting what we hear best. His comparison is painting a picture with perfect colors for a viewer who is color-blind, while getting every shape wrong. You have perfected the part they can't see and spoiled the part they can.

1.2 #7 Head movement matters more than individual ear shape.

> "3-D" headphones fail mainly because they ignore head movement ("We're all basically shaped the same"); at Harman, open-back headphones with a head tracker and DSP were indistinguishable from a real speaker. RARE
>
> **Sources:** DIYMA 2016-05-22 · DIYMA 2016-06-23

In plain English: "3-D" headphones use processing to try to make sound seem to come from outside your head. Many people believe they fail because everyone's ears are shaped differently. He says they fail mainly because they ignore head movement: "We're all basically shaped the same." He worked at Harman, the company behind JBL. There, open-back headphones combined with a head tracker (a sensor that follows how you turn your head) and DSP processing could not be told apart from a real speaker.

1.2 #8 What you see moves the center image.

> Moving an iPod controller from left of center to the middle of his dash improved his IASCA center-image scores; ears work with eyes, and early reflections confirm what the eyes see. RARE
>
> **Sources:** DIYMA 2013-05-03 · OFCAP Ep. 38 @39:05

In plain English: IASCA is a car-audio sound-quality competition where judges score things like how well the center image is placed. He says that when he moved an iPod controller from the left of center to the middle of his dash, his center-image scores improved. His explanation is that our ears work together with our eyes, and the early reflections confirm what the eyes see. In other words, the brain uses what it sees as well as what it hears to decide where a sound is.

1.3 Reflections: what makes a car sound like a car

What this part is about: A car is a small box of glass, plastic and fabric, and sound bounces off all of it. This part explains what those bounces (reflections) do to the sound you hear and the image you see, why EQ and delay can't remove them, and how speaker placement can keep them from doing harm.

1.3 #1 A car has no reverberant field, so you mostly hear the speaker's power response.

> Every reflection arrives within ~20 ms and fuses with the direct sound into roughly the average of the speaker's output over all angles (its "power response"); there are no late reflections. So a speaker swap often changes little, and EQ is the main lever. RARE
>
> **Sources:** DIYMA 2009-12-06 · DIYMA 2017-10-28 · DIYMA 2017-08-27

In plain English: In a large room or concert hall, sound keeps bouncing around for a while after the music stops. That is reverberation, and it includes late reflections that arrive well after the direct sound. He says a car is far too small for this. Every reflection arrives within about 20 ms (twenty thousandths of a second) and blends with the direct sound. What you hear is roughly the average of everything the speaker sends out in all directions, called its "power response". So swapping one speaker for another often changes less than people expect, and EQ (turning particular frequencies up or down) is the main tool for shaping the sound.

1.3 #2 Cross-car reflections near your ears are what make a car "sound like a car".

> Right-channel sound bouncing off surfaces by your left ear (and vice versa) is acoustic crosstalk that narrows the image; worst are the far speaker off the near side glass and both speakers off the windshield. Home listeners call car systems "mono" for this; neither horns nor EQ remove it. RARE
>
> **Sources:** DIYMA 2010-01-03 · DIYMA 2011-11-17 · CMA Networks 2023 @38:04

In plain English: Crosstalk means one channel leaking into the other. Here, sound from the right speaker bounces off surfaces near your left ear, and the reverse, so each ear gets some of the wrong side's sound. He says this narrows the stereo image. The worst cases are the far speaker's sound bouncing off the side window next to you, and both speakers' sound bouncing off the windshield. He says this is why people used to home stereo call car systems "mono", meaning everything seems to come from one place. Horn speakers (which steer sound in a controlled direction) don't remove it, and neither does EQ.

1.3 #3 Reflections show up as image spread and wandering, not as echoes.

> Each reflection acts as another source; if its response differs from the direct sound, images land in different places per frequency. Below 1 kHz an out-of-phase reflection can weaken one side so pans stick near the speakers (a classic stuck drumbeat in pan tests); an A-pillar mid acts like four mids. RARE
>
> **Sources:** DIYMA 2013-11-20 · DIYMA 2022-02-23 · DIYMA 2020-10-07

In plain English: Many people expect reflections to cause echoes. He says that in a car they show up instead as an image that spreads out or wanders. Each reflection acts like another speaker. If the reflected sound has a different tonal balance from the direct sound, images land in different places at different pitches. Below 1 kHz, a reflection that arrives out of phase can weaken one side.

Then, when a test track pans a sound from one side to the other, it sticks near the speakers instead of moving smoothly. His classic example is a drumbeat that gets stuck in a pan test. Since each strong reflection acts as another source, he says a midrange in an A-pillar (the post beside the windshield) acts like four midranges.

1.3 #4 No EQ or delay can fix a reflection comb.

> The processor sits before the speaker, so any EQ or delay applies to the reflection too ("that's the law"), and reflections don't change delay settings. Box peaks are EQ-able, reflection dips aren't. If a dip can't be moved, copy it into the other side: the image improves though the dip stays. RARE
>
> **Sources:** Time Alignment Part 2 · DIYMA 2021-03-26 · DIYMA 2021-11-06

In plain English: The processor (DSP) sits before the speaker in the signal chain. Any EQ or delay you apply changes what the speaker plays. The reflection is just a bounced copy of that sound, so it gets exactly the same change: "that's the law". The reflection still arrives later by the same amount, so the dip it causes stays. He says peaks caused by a speaker's box can be fixed with EQ, but dips caused by reflections can't. His workaround applies when you can't move the dip by moving the speaker. Copy the same dip into the other side with EQ. The dip remains, but the image improves because left and right now match.

1.3 #5 Predict reflection dips with a tape measure before you build.

> A dip falls where the reflected path is half a wavelength longer than the direct one: a mid 6" from a surface (~1 ft extra) dips near 500 Hz; a 2-ft path near 250 Hz; a 600 Hz null means ~11" extra. Never put a null inside a driver's passband: push mids into corners (null above their range) and move tweeters away (null below theirs). RARE
>
> **Sources:** Audiofrog blog Jun 2021 (Dispersion and reflection) · DIYMA 2020-10-07 · DIYMA 2021-11-06

In plain English: A wavelength is the length of one full sound wave: low notes have long waves and high notes short ones. He says a dip appears at the pitch where the bounced path is half a wavelength longer than the direct path. At that pitch the reflection arrives exactly opposite to the direct sound. His examples: a midrange 6 inches (15 cm) from a surface gets about 1 foot (30 cm) of extra path and dips near 500 Hz. A 2-foot (60 cm) extra path dips near 250 Hz. A dip at 600 Hz means about 11 inches (28 cm) of extra path. His rule is never to put a dip inside the range a speaker plays. So in practice: push midranges into corners, so the dip lands above their range, and move tweeters away from surfaces, so the dip lands below theirs.

1.3 #6 Mount a driver in a surface, not in front of it.

> A surface used as the baffle stops being a troublesome reflector: in-door mounting removes the door panel as a null source, in-dash removes the dash, and deep in a corner the return paths are so short you get only gain. A pod standing off a baffle adds exactly the dip you're avoiding. RARE
>
> **Sources:** DIYMA 2011-04-15 · DIYMA 2020-10-07

In plain English: A baffle is the panel a speaker is mounted in. He says that if a surface itself is the baffle, with the speaker set into it, it stops being a reflector that causes trouble. Mounting in the door removes the door panel as a cause of dips. Mounting in the dash removes the dash. Deep in a

corner, the bounced paths are so short that you only get extra loudness (gain), not dips. A pod is a housing that holds a speaker out in front of a surface. He says a pod standing off a baffle adds exactly the kind of dip you were trying to avoid.

1.3 #7 Distance is the lever: reflections drop 6 dB per doubling, and one next to your ear hurts as much as one next to the speaker.

> Carpet absorbs the top octaves and glass reflects them; the driver's own window causes as much response error as the windshield, and no pattern control changes that. RARE
>
> **Sources:** DIYMA 2012-09-06 · DIYMA 2016-04-23

In plain English: He says a reflection gets about 6 dB quieter each time the distance it travels doubles, so distance is the main way to weaken it. A surface right beside your ear does as much harm as one right beside the speaker. Carpet absorbs the top octaves, the highest treble, while glass reflects them. His example: the window next to the person driving, beside their ear, causes as much error in the frequency response as the windshield next to the speaker. No pattern control, meaning speaker designs that aim sound in a set direction, changes that.

1.3 #8 Dash mats, foam and films don't control midrange reflections.

> Absorbers start working at about a quarter wavelength, so ~200 Hz would need ~1.5 ft of absorber; a dash mat "does NOTHING" in the lower midrange and foam just stops rattles. Real absorbers are membranes (what mass-loaded vinyl does in walls) or fiber stuffing. RARE
>
> **Sources:** DIYMA 2013-09-14 · DIYMA 2022-01-14

In plain English: An absorber is soft material that soaks up sound instead of bouncing it back. He says an absorber starts working only once it is about a quarter of a wavelength thick. At 200 Hz that means about 1.5 feet (about 45 cm) of material. So a dash mat "does NOTHING" in the lower midrange, and foam just stops rattles. Real absorbers, he says, are membranes or fiber stuffing. A membrane absorber is a heavy, flexible sheet, which is what mass-loaded vinyl does inside walls.

1.3 #9 A 250 Hz "resonance" is usually a reflection peak to EQ, not a panel to deaden.

> A true resonance is mass and spring trading energy; a lone peak with dips either side is usually a cabin dimension reinforcing that frequency. Play a 250 Hz tone: if nothing buzzes, EQ it down. RARE
>
> **Sources:** DIYMA 2021-09-07

In plain English: A true resonance, he says, is mass and spring trading energy back and forth, like a panel vibrating. Many people hear a peak around 250 Hz, decide a panel is resonating, and add sound deadening (material stuck to panels to stop vibration). He says a single peak with dips on either side is usually a cabin dimension reinforcing that frequency: sound arriving from two distances and adding up. His test: play a 250 Hz tone. If nothing buzzes or rattles, EQ the peak down.

1.3 #10 Two strategies for reflections, and he picks "spread the chaos around".

> Either minimize reflections with horns or big diaphragms (pattern control), or use each driver only where its dispersion is wide, so reflections share the direct sound's response and one EQ fixes both. He prefers the second (drivers exist, mounting is easier, EQ works); neither fixes reflections near

> the ears. RARE
>
> **Sources:** DIYMA 2010-01-03 · DIYMA 2013-08-10

In plain English: He describes two ways to deal with reflections. The first is to minimize them using pattern control, meaning speakers that send sound only in a chosen direction, such as horns or large diaphragms (big speakers beam more). The second is to use each speaker only in the range where it still spreads sound widely. Then the reflections have the same tonal balance as the direct sound, and one EQ setting fixes both. He prefers the second, which he calls "spread the chaos around", because suitable speakers exist, mounting is easier, and EQ works. He notes that neither approach fixes reflections near your ears.

1.3 #11 Kill the reflections and the car sounds like headphones: aim for a pleasing room, not no room.

> The least reflective car he heard was "very accurate" but "missed the sense of space"; stereo in an anechoic chamber was horrible. Inverting the car's impulse response works only for a head that never moves, but it is a good base for adding a virtual room. RARE
>
> **Sources:** DIYMA 2013-09-15 · DIYMA 2017-07-05 · DIYMA 2017-07-04

In plain English: Many people think the ideal car would have no reflections at all. He says the least reflective car he has heard sounded "very accurate" but "missed the sense of space", much like headphones. He has also heard stereo in an anechoic chamber, a test room with no echoes at all, and says it was horrible. So the aim is a pleasing room, not no room. An impulse response is a measurement of how the car changes a single sharp click. Some people propose "inverting" it with a processor to cancel the reflections. He says that works only for a head that never moves, but it is a good starting point for adding a virtual room, an artificial sense of space created by the processor.

1.3 #12 Windshield reflections are not ambience: a car has "spread", not early and late reflections.

> They arrive far too early to suggest a stage or room, so they just unfocus the image and leave a jagged comb; cabin reflections can't add depth either. There is "no ambience in a car unless you create it" with rear speakers and delay. RARE
>
> **Sources:** DIYMA 2013-12-15 · DIYMA MS-8 FAQ p3 · DIYMA 2022-02-23

In plain English: In a concert hall, early reflections and late reflections arriving at different times give a sense of the room and of space. That sense of space is called ambience. He says a car's windshield reflections arrive far too early to suggest a stage or a room. They just blur the image, which he calls "spread", and leave a jagged comb of dips. Reflections inside the cabin can't add depth either. There is "no ambience in a car unless you create it", which you do with rear speakers and delay.

1.4 Dispersion and directivity: the property that decides placement

What this part is about: Dispersion means how widely a speaker spreads its sound. This part explains why that depends mainly on the speaker's size compared with the length of the sound waves it plays. He then uses that to decide which size of speaker should play which range, where

the crossovers should go, and why many cars need three speakers per side rather than two.

1.4 #1 Speakers are floodlights, not flashlights: dispersion is set by diameter versus wavelength.

> Every driver radiates everywhere at wavelengths long relative to its size and beams at short ones. Cone material, phase plugs, ring radiators and "diffusers" barely change it: two 1" tweeters differ off-axis only above ~3 kHz, and a 15 kHz dip at 60° doesn't matter in a car. RARE
>
> **Sources:** DIYMA 2010-07-11 · DIYMA 2014-06-01 · DIYMA 2021-11-23

In plain English: A floodlight lights up everything around it; a flashlight throws a narrow beam. He says every speaker sends sound in all directions at pitches whose waves are long compared with its size, and "beams" (sends sound mostly straight ahead) at pitches whose waves are short. So the speaker's diameter compared with the wavelength sets its dispersion. Many people believe cone material, phase plugs (the bullet shape in the middle of some speakers), ring-radiator tweeters or "diffusers" widen a speaker's spread a lot. He says they barely change it. "Off-axis" means heard from an angle rather than straight in front. Two different 1-inch tweeters, he says, differ off-axis only above about 3 kHz. A dip at 15 kHz heard 60° off to the side doesn't matter in a car.

1.4 #2 His dispersion rule in numbers: driver circumference ÷ wavelength.

> Below 1: same response on- and off-axis; ~2: narrowing starts; ~3: matches only to ~30°; ~4: ~15°; ~10: on-axis only. Working figures: a 6" is omnidirectional below ~300–500 Hz, a 3" below ~3 kHz, a 4" below ~6 kHz; a 1" tweeter beams only above ~10 kHz. RARE
>
> **Sources:** DIYMA 2014-03-20 · Audiofrog blog Jun 2021 · DIYMA 2013-12-15

In plain English: Circumference is the distance around the speaker's cone, about 3.14 times its diameter. His rule divides that by the wavelength of the sound being played. Below 1, the speaker sounds the same straight in front and off to the side. Around 2, the spread starts to narrow. Around 3, the sound matches only within about 30° of straight ahead. Around 4, only within about 15°. Around 10, it is good only straight ahead. His working figures follow from this. A 6-inch speaker spreads evenly in all directions (omnidirectional) below about 300–500 Hz. A 3-inch does so below about 3 kHz, and a 4-inch below about 6 kHz. A 1-inch tweeter beams only above about 10 kHz.

1.4 #3 Off-axis response matters more in a car than in a room.

> Surfaces are closer, so reflections are louder and you hear on- and off-axis output combined; Harman's room-directivity research applies even more strongly in cars. RARE
>
> **Sources:** DIYMA 2009-07-10 · DIYMA 2015-05-07

In plain English: "On-axis" is the sound straight in front of a speaker; "off-axis" is the sound it sends out at an angle. In a car, the surfaces are much closer to the speakers than the walls of a room are. So, he says, the reflections are louder, and what you hear is the on-axis and off-axis output combined. Harman, the company he worked for, is known for research on how a speaker's off-axis sound affects how it sounds in a room. He says that research applies even more strongly in cars.

1.4 #4 Big woofer + small tweeter = an off-axis hole at the crossover = far-side bias.

> A 6.5–8" with a 1" dome can be flat on-axis yet leave "a big hole" off-axis; aiming only moves the hole between direct and reflected sound. Aimed on-axis, leave the dip; flat in the door, EQing it makes the passenger seat worse. A 6" + 1" is "a bad deal"; a 3-way fixes it. RARE
>
> **Sources:** Crutchfield · DIYMA 2013-03-10 · DIYMA 2021-11-30

In plain English: A 6.5–8 inch woofer is already narrowing its spread by the frequency where it hands over to a 1-inch dome tweeter. So the pair can measure flat straight in front but leave "a big hole" in the sound sent off to the sides around the crossover. Far-side bias means the image leaning toward the speaker on the far side of the car. He says aiming the speakers only moves the hole between the direct sound and the reflected sound. If the speakers are aimed at you, leave the dip alone. If they sit flat in the door, you can EQ the dip, but that makes the passenger seat worse. His reason: the near speaker needs more boost, which is the opposite of what the passenger seat needs. He calls a 6-inch plus 1-inch pairing "a bad deal" and says a 3-way (adding a midrange) fixes it.

1.4 #5 The target is a smoothly narrowing pattern, not identical on- and off-axis response.

> Directivity index means how much more a speaker sends forward than its average; it should rise gradually, with no holes or peaks near crossovers (off-axis equal to a flat on-axis sounds too bright). Home crossover rules apply in cars because they make reflected sound resemble direct sound. RARE
>
> **Sources:** DIYMA 2009-07-10 · DIYMA 2013-09-17 · DIYMA 2021-09-02

In plain English: Many people believe a speaker should sound exactly the same straight in front and off to the side. He says the target is different: a pattern that narrows smoothly as the pitch rises. The point defines directivity index. He says it should rise gradually with frequency, with no holes or peaks near the crossovers. If the off-axis sound were equal to a flat on-axis sound, the result would sound too bright. The usual rules for designing home-speaker crossovers apply in cars too, he says, because they make the reflected sound resemble the direct sound. He adds that this makes EQ easier, because the measuring microphone can't tell direct sound from reflected sound.

1.4 #6 Smooth directivity needs a waveguide or a 3-way, and cars need the 3-way.

> Studio monitors can be 2-ways because the engineer sits ~3 ft away with walls ~10 ft off; in a car the reflectors touch the speakers. A waveguide big enough to match a 6" (his home GB60/GB15 one is ~5" effective) won't fit a car. RARE
>
> **Sources:** DIYMA 2022-01-05 · DIYMA 2008-12-02 · DIYMA 2015-12-21

In plain English: A waveguide is a shallow horn shape around a tweeter that narrows its spread at the low end, so it can match a bigger woofer. A 3-way splits the sound among three speakers per side (woofer, midrange and tweeter) instead of two. He says studio monitors can be 2-ways because the recording engineer sits about 3 feet (about 1 m) away, with walls about 10 feet (about 3 m) off. In a car, the reflecting surfaces are right against the speakers. A waveguide big enough to match a 6-inch woofer won't fit in a car. For scale, the one he uses at home with the GB60 woofer and GB15 tweeter (Audiofrog products) is about 5 inches effective. So, in a car, he says the answer is the 3-way.

1.4 #7 A waveguide should narrow the tweeter only at the crossover.

> Narrowing its low end to meet the mid is fine if the curve stays smooth; narrowing its top end to match a beaming woofer leaves a power-response hole that "NEVER sounds good". The JBL 660GTi waveguide was designed for ~35° off-axis door mounting. RARE
>
> **Sources:** DIYMA 2013-03-22 · DIYMA 2014-04-12

In plain English: He says a waveguide should narrow the tweeter's spread only around the crossover. Narrowing its low end so it meets the midrange is fine, as long as the change stays smooth. Narrowing its top end, to match a woofer that is already beaming, leaves a hole in the power response (the total sound sent in all directions, which is mostly what you hear in a car; see 1.3 #1). He says that "NEVER sounds good". He notes that the JBL 660GTi waveguide was designed for mounting in a door at about 35° off-axis.

1.4 #8 Design crossovers for the angle each driver is heard at, in the car.

> His 4"-plus-tweeter passive (~2.7 kHz) was designed from the mid measured ~30° off-axis; an anechoic design moved into a car is not the same thing. Same plane matters more than same axis. RARE
>
> **Sources:** Time Alignment Part Three · DIYMA 2014-06-03

In plain English: A passive crossover is a small circuit of capacitors and coils that splits the music between speakers. He designed his crossover for a 4-inch midrange plus tweeter (crossing at about 2.7 kHz) using the midrange measured about 30° off-axis, the angle at which it is heard in the car. He says a crossover designed in an anechoic chamber (a test room with no reflections) and then moved into a car is not the same thing. So in practice: design for the angle at which each speaker is actually heard. He adds that mounting the two drivers in the same plane matters more than putting them on the same axis. Same plane means their faces sit on the same surface; same axis means they are lined up along one line.

1.4 #9 Choose crossover points by dispersion and driver safety, not "as low as possible".

> First keep the smaller driver safe (tweeter at ~2× its resonance), then hand off where the larger one still disperses widely, below any cone breakup peak. His pairings: 1.5" tweeter at ~1.5 kHz with a 6" ("the killer app" 2-way); 1" near 2.5 kHz, steep; 6" to small mid at 400 Hz–1 kHz; 3" to tweeter at 3.5–4 kHz. RARE
>
> **Sources:** OFCAP Ep. 19 @1:41:04 · DIYMA 2014-03-04 · DIYMA 2012-09-06

In plain English: Many people believe you should cross over to the tweeter "as low as possible". He says to choose the point by dispersion and driver safety instead. First, keep the smaller speaker safe: cross a tweeter at about twice its resonance, the pitch at which it naturally vibrates most easily. Then hand off where the larger speaker still spreads sound widely, and below any cone breakup peak, a spike that appears where the cone stops moving as one piece. His pairings: a 1.5-inch tweeter at about 1.5 kHz with a 6-inch woofer, which he calls "the killer app" 2-way. A 1-inch tweeter near 2.5 kHz, with a steep filter. A 6-inch to a small midrange at 400 Hz–1 kHz. A 3-inch midrange to a tweeter at 3.5–4 kHz.

1.4 #10 Real pattern control is impractical in a car below ~1 kHz.

> Using natural beaming would take ~15" midbass, 10" mids and 4" tweeters, or horns "and a school bus to put them in"; it "completely falls apart in the midbass". RARE
>
> **Sources:** DIYMA 2013-02-04 · DIYMA 2010-02-16

In plain English: Pattern control means steering where a speaker's sound goes, so less of it hits nearby surfaces. To do that using speakers' natural beaming, he says you would need about 15-inch midbass speakers, 10-inch midranges and 4-inch tweeters. The alternative is horns "and a school bus to put them in". He says the approach "completely falls apart in the midbass". This ties in with 1.3 #10, where he prefers using each speaker only where it spreads sound widely.

1.4 #11 The point of a small midrange is crossover placement, not "all vocals from one driver".

> A 2.5" bridges a 6" and a 1" so every crossover sits where dispersion is wide; running a tiny mid down to 200 Hz only costs distortion and power handling. Keeping vocals in one driver is "audio mythology at its worst". RARE
>
> **Sources:** DIYMA 2008-11-30 · DIYMA 2014-03-21

In plain English: A small midrange, such as a 2.5-inch, sits between the woofer and the tweeter. Many people believe its purpose is to play all of the vocals from one speaker. He calls that idea "audio mythology at its worst". He says its real job is to bridge a 6-inch woofer and a 1-inch tweeter, so that every crossover sits where both speakers still spread sound widely. Running a tiny midrange down to 200 Hz to cover more of the voice only costs distortion and power handling (how much power it can take without damage).

1.5 Aiming: only where a driver beams

What this part is about: Pointing ("aiming") a speaker at the listener only matters at frequencies where that speaker sends its sound out in a narrow beam. His view is that in a car this is mostly true of the tweeter, so aiming woofers and midranges achieves little.

1.5 #1 In a 3-way, aim "Just the tweeter."

> A 6" below ~300 Hz and a 3" below ~3 kHz are identical from 0 to 90°; for a 2" pillar mid crossed at 2 kHz, aiming "HAS NO EFFECT WHATSOEVER" on reflections. Turn tweeters in a little; a gauge cluster in a mid's path changes little and EQ fixes it. RARE
>
> **Sources:** Audiofrog blog Jun 2021 · DIYMA 2016-04-22 · DIYMA 2021-10-29

In plain English: A 3-way system splits the music among three kinds of speaker: a woofer or midbass for low notes, a midrange for the middle, and a tweeter for the highs. Any speaker spreads low notes evenly in every direction and only starts to "beam" (send its sound mostly straight ahead) as the pitch rises. He says a 6" (about 150 mm) speaker below about 300 Hz, and a 3" (about 75 mm) speaker below about 3 kHz (3,000 vibrations per second), sound the same whether you are straight in front (0°) or fully to the side (90°). So turning them changes nothing. For a 2" pillar midrange crossed at 2 kHz (it hands the notes above 2 kHz to the tweeter), he says aiming has no effect at all on the reflections it makes off the glass and dash. Only the tweeter plays high enough

for aiming to count. So in practice: turn the tweeters in a little. If a gauge cluster (the instrument panel) sits in a midrange's path, he says it changes little and EQ (adjusting the level of each frequency band) fixes it.

1.5 #2 Door midbass aiming doesn't matter at all.

> From 80 to 200 Hz a 6" radiates everywhere ("not one bit — zip — zilch — nada"); what matters is arrival time and the sub-to-midbass blend. RARE
>
> **Sources:** DIYMA 2010-02-14

In plain English: Midbass is the upper-bass range, roughly the punch of a kick drum, usually played by a 6" (about 150 mm) speaker low in each door. He says that from 80 to 200 Hz a 6" speaker sends sound out equally in every direction, like ripples spreading from a pebble dropped in a bathtub, so which way it faces makes no difference at all. What matters instead is arrival time (when its sound reaches your ears compared with the other speakers) and how smoothly it blends with the subwoofer. So in practice: don't spend effort angling door woofers; spend it on timing and on the sub-to-midbass blend.

1.5 #3 Aiming off-axis is just slow, risky "acoustic EQ", and it adds crosstalk.

> Pointing a speaker away moves its highs into the reflections, which is why A-pillar and into-the-windshield mounting give a narrower stage than door or sail-panel tweeters. A friend who liked "head between the tweeters" was just closer to them (louder). RARE
>
> **Sources:** DIYMA 2011-03-26 · DIYMA 2014-03-30 · DIYMA 2021-10-29

In plain English: "Off-axis" means the speaker points away from you, so you hear it from the side. Because a speaker beams its highs, turning it away lowers the highs you hear directly and sends them toward the glass and panels, so you hear them mostly as reflections. He calls this a slow, risky "acoustic EQ": a tone control you set by physically moving the speaker. It also adds crosstalk, meaning sound meant for one side spreading to the other, for example by bouncing across the windshield. That, he says, is why speakers in the A-pillars (the posts either side of the windshield) or aimed into the windshield give a narrower stage (the band of sound you seem to hear in front of you) than tweeters in the doors or sail panels (the small triangles where the side mirrors mount). A friend who liked having his "head between the tweeters" was, in his view, simply closer to them, so they were louder.

1.5 #4 Tweeter rule: keep both listeners inside a ~±30–35° window and stop fussing.

> 0° versus 15° makes essentially no difference; turn sail-panel tweeters slightly toward the rear of the car. Laser aiming is "nonsense" unless you "strap your head to the headrest".
>
> **Sources:** DIYMA 2011-04-15 · DIYMA 2012-04-29 · DIYMA 2015-12-20

In plain English: Picture the tweeter's beam as a cone spreading out from it. He says that if both the driver and the passenger sit within about 30 to 35 degrees either side of where the tweeter points, that is good enough, and you can stop fussing. Pointing it straight at you (0°) versus a little off (15°) makes essentially no difference. His specific tip: turn sail-panel tweeters slightly toward the rear of the car. Aiming tweeters with a laser at an exact head position is "nonsense", he says, unless you "strap your head to the headrest", since a real listener's head moves.

1.5 #5 A-pillar tweeters aimed straight back or at the listener just trade one error for another.

> Straight back matches level between seats but the center wanders; aimed at the listener matches response but adds near-side bias. With time alignment available, move them to the sail panels.
>
> **Sources:** DIYMA 2011-11-05

In plain English: A-pillar tweeters sit on the posts either side of the windshield. He describes two common ways to point them, each with its own flaw. Aimed straight back toward the rear of the car, both seats hear similar loudness, but the center image (the sound that should seem to come from the middle, like a lead singer) wanders. Aimed at the listener, the tone matches better, but you get near-side bias: the speaker closer to you dominates and pulls the sound toward your side. If you have time alignment (a processor that delays the nearer speakers so all the sound arrives together), his advice is to move the tweeters to the sail panels instead.

1.5 #6 Crossfiring A-pillar tweeters across the dash is "a band-aid for a 2-seat car".

> The near tweeter loses its top octave off-axis, roughly matching the far one that is ~6 dB down by distance: better for both seats than aiming at each, worse than a one-seat tune. RARE **Conflict:** a 2015 Harman/JBL/Infinity tip sheet (authorship unconfirmed) recommends it, and the Bose Fiat 500 he praised for two-seat imaging crossfires.
>
> **Sources:** DIYMA 2012-04-07 · Harman tip sheet 2015 · DIYMA 2015-12-07

In plain English: Crossfiring means pointing each A-pillar tweeter across the dash toward the opposite seat. Take the driver: the near tweeter now points away from him, so he hears it off-axis and it loses its top octave (the highest band of treble; an octave is a doubling of pitch). The far tweeter points at him but is about 6 dB quieter simply because it is farther away (6 dB is a clearly audible drop). The two losses roughly balance, so both sides sound more alike. He calls this "a band-aid for a 2-seat car": a patch that works better for both seats than aiming each tweeter at its own listener, but worse than tuning for one seat. The [Conflict] tag notes the other side: a 2015 tip sheet from Harman/JBL/Infinity (whose author is unconfirmed) recommends crossfiring, and the Bose system in the Fiat 500, which he praised for imaging in both seats, crossfires.

1.5 #7 He co-invented an aimable flush swivel tweeter at Infinity.

> The premise: highs are directional and car locations often poor, so a flush tweeter can be aimed toward the listener (up to ~27° in the figures), consistent with his later view that aiming is a tweeter-only tool. **Era:** Infinity, filed 1997
>
> **Sources:** US 5,859,917

In plain English: While at Infinity (a speaker brand owned by Harman), he was a co-inventor on US patent 5,859,917, filed in 1997. It covers a tweeter mounted flush (level with the panel surface, not in a pod) that can swivel in its mount so it can be pointed toward the listener, by up to about 27° in the patent's drawings. The idea was that high notes are directional and the spots available in cars are often poor ones. The point notes that this fits his later view that aiming is a tool for tweeters only. This point relates to a specific product and period of his career (Infinity, 1997).

1.6 Where each driver goes: doors, sail panels, A-pillars, kicks, dash, corners

What this part is about: Where to physically mount each speaker in a car: doors, sail panels, A-pillars, kick panels, the dash and the corners. His recurring theme is that nearby glass and panels reflect sound and shape what you hear, so every spot is a trade-off between width, height, tone and effort.

1.6 #1 His default layout: a 3-way in the doors, small mid high, tweeter in the sail panel.

> 6" midbass low in the door, a 2.5–4" mid high ("in front of the door handle"), tweeter in the sail panel. Every Audiofrog demo car (through the 2023 Volvo C30) and his MS-8 development cars used it: the glass becomes the baffle, the windshield is farther away, the stage is widest, and "No surprises." RARE
>
> **Sources:** DIYMA 2013-03-11 · CMA Networks 2023 @17:01 · Absolute Electronix 2023 @51:00

In plain English: This is his go-to setup. A 3-way means three speakers per side: a 6" (about 150 mm) midbass low in the door, a small 2.5–4" (about 65–100 mm) midrange high in the door, "in front of the door handle", and the tweeter in the sail panel (the small triangle by the side mirror). He used it in every Audiofrog demo car (Audiofrog is his speaker company), through the 2023 Volvo C30, and in the cars used to develop the JBL MS-8 processor. His reasons: the door glass becomes part of the baffle (the surface a speaker is mounted in, which guides its sound forward); the windshield is farther away; the stage comes out widest; and "No surprises." Many people believe the A-pillars are the best place for mids and tweeters; this point goes against that.

1.6 #2 "The key is in getting a small midrange high in the door."

> A 2.5" high in the door can outperform a 4" in the wrong spot; a GB25 in a door panel needs no enclosure, ideally ahead of the door pull. Choose door versus elsewhere by whether you value width or height more, since width follows placement. RARE
>
> **Sources:** DIYMA 2014-03-04 · DIYMA 2017-10-31 · OFCAP Ep. 38 @35:03

In plain English: He says "The key is in getting a small midrange high in the door." A 2.5" (about 65 mm) midrange placed high in the door can beat a 4" (about 100 mm) one in the wrong spot. The GB25, Audiofrog's 2.5" midrange, needs no enclosure (box) when mounted in a door panel, ideally ahead of the door pull (the handle you grab to close the door). Choosing between the door and other spots, he says, comes down to whether you care more about stage width or stage height, because width follows where the speakers physically are.

1.6 #3 Midbass always sits in a corner in a car, so design and EQ for it.

> Low door and kick are corners: the baffle adds ~6 dB and the corner ~6 dB more below ~500 Hz, which is why the GB60 is voiced flat and low-Q rather than "full". A woofer in the door's top front corner makes left and right responses much more alike; with a sub you don't need an 8". RARE
>
> **Sources:** Audiofrog blog Jun 2021 · DIYMA 2017-08-17 · DIYMA 2021-12-03

In plain English: A corner is where two or three surfaces meet. The low part of a door and the kick panel (the side panel in the footwell) are both corners. Nearby surfaces reinforce a speaker's lower notes: he says the baffle adds about 6 dB, and the corner about 6 dB more, below roughly 500 Hz.

(6 dB is roughly double the sound pressure.) Because the car already adds this bass, Audiofrog's GB60 midbass is "voiced" (tuned by design) flat and low-Q, meaning smooth rather than bumped up to sound "full". He adds that a woofer in the door's top front corner makes the left and right sides measure much more alike, and that if you have a subwoofer you don't need an 8" (about 200 mm) midbass.

1.6 #4 Cross door midbass high, 400 Hz–1 kHz.

> He always does and has "never had" a 400–600 Hz console cancellation; he endorsed 80–800 Hz lower-door midbass with an upper-door mid, and a 9" door (70–600 Hz) plus 3.5" kick 3-way. Mini Cooper fix: add a 3" and cross the factory 6" at 1 kHz.
>
> **Sources:** DIYMA 2014-03-30 · DIYMA 2014-04-02 · DIYMA 2010-01-03

In plain English: A crossover is the filter that decides which speaker plays which notes; the crossover point is where one speaker hands over to the next. He crosses the door midbass high, somewhere between 400 Hz and 1 kHz, so the door woofer plays up to that point before the midrange takes over. When a forum member worried about a cancellation dip from the console at 400–600 Hz, he said he always crosses that high and has "never had" that problem. He has endorsed a lower-door midbass playing 80–800 Hz with a midrange high in the door, and a 3-way with a 9" (about 230 mm) door woofer playing 70–600 Hz plus a 3.5" kick-panel speaker. For a Mini Cooper, his fix was to add a 3" (about 75 mm) speaker and cross the factory 6" over at 1 kHz.

1.6 #5 The sail panel (mirror triangle) is his tweeter location.

> It's farther from the windshield, the door and glass act as its baffle, it gives a wider stage than dash or pillars, and it's easy ("Velcro them on and recalibrate"). About 1.5" from the glass puts the reflection dip near 2.2 kHz, below most of the tweeter's range. Tweeters in kicks? Move them to the sails. RARE
>
> **Sources:** DIYMA 2011-11-05 · DIYMA 2012-04-08 · DIYMA 2017-10-31

In plain English: The sail panel is the small triangle at the front of the door window where the mirror mounts. He prefers it for tweeters because it is farther from the windshield, the door and glass act as its baffle, it gives a wider stage than the dash or pillars, and it's easy: "Velcro them on and recalibrate" (re-run the processor's setup). Background: when a speaker sits near glass, the bounced sound arrives slightly late and cancels the direct sound at one pitch, leaving a dip. He says about 1.5" (about 38 mm) from the glass puts that dip near 2.2 kHz, below most of what a tweeter plays. If your tweeters are in the kick panels, his advice is to move them to the sails.

1.6 #6 A-pillars are "a HORRIBLE place for a midrange"; their one benefit is height.

> A pod hanging a foot or so from windshield, dash and side glass spreads the image over all three and combs the whole midrange: a 2-ft reflection path digs a hole near 250 Hz (the "mid won't play below 250" complaint). Baffle-less tweeter pods there give ~4 apparent sources per side. RARE Conflict:
>
> Audiofrog's own copy calls GB10+GB25+GB60 "the ultimate active system for A-pillars", and the GB25 is designed for pillar pods.
>
> **Sources:** DIYMA 2013-03-10 · DIYMA 2021-08-26 · GB60 page

In plain English: He calls the A-pillars "a HORRIBLE place for a midrange", with height as their one benefit. A pod hanging about a foot (30 cm) from the windshield, dash and side glass reflects off all three, so the image spreads across all three. Those reflections also cause comb filtering across the

whole midrange: when a sound mixes with a slightly delayed copy of itself, a row of dips appears, like the teeth of a comb. He says a reflection path of about 2 feet (60 cm) digs a hole near 250 Hz, which explains the common complaint that a pillar "mid won't play below 250". Tweeter pods with no baffle there create about 4 apparent sound sources per side. The [Conflict] tag: his own company's product text calls the GB10+GB25+GB60 set "the ultimate active system for A-pillars", and the GB25 is designed for pillar pods.

1.6 #7 A-pillar pods are enclosures: size the box to the driver, not the biggest driver to the pillar.

> A big infinite-baffle driver in a tiny pod loses bass and peaks before rolloff, like a sub in a too-small box. The GB25 was designed so "you cannot build an enclosure too small": with a 200 Hz LR4 high-pass, 0.16 L, 0.25 L and huge boxes measure almost alike. RARE
>
> **Sources:** DIYMA 2017-10-01 · DIYMA 2017-07-02 · CMA Networks 2021 @17:00

In plain English: A pod on a pillar is really a small box, and a speaker's bass depends on the box it's in. Some speakers are "infinite-baffle" types, made to work with a large open space behind them (such as a door) rather than a small sealed box. Put a big one of these in a tiny pod and, he says, it loses bass and gets a peak just before its low end falls away, like a subwoofer in a box that is too small. So pick a driver that suits the pod, rather than the biggest driver that fits the pillar. Audiofrog designed the GB25 so that "you cannot build an enclosure too small": with a 200 Hz LR4 high-pass (a steep filter that removes the notes below 200 Hz), boxes of 0.16 L, 0.25 L (about one cup) and very large ones measured almost the same.

1.6 #8 Windshield corner: push the mid tight into it; never hang a pod about a foot away.

> Tucked into the dash–windshield–pillar corner, the reflectors are so close that the first null moves above the mid's range, which is why factories use it; its position relative to the glass matters more than mid-to-tweeter spacing. RARE **Changed view:** 2012 and March 2013 said keep speakers (mids especially) away from that junction; from August 2013, and firmly in 2021, tight in the corner is best and hanging near it is the problem.
>
> **Sources:** DIYMA 2012-09-06 · DIYMA 2013-03-20 (old view) · DIYMA 2013-08-10 · DIYMA 2021-12-24

In plain English: Background: the closer a reflecting surface is to a speaker, the higher the pitch of the first cancellation dip (the "first null") it causes. He says that if you push a midrange tight into the corner where the dash, windshield and pillar meet, the surfaces are so close that the first null moves above the range the mid plays. That, he says, is why carmakers use the spot. Hanging a pod about a foot (30 cm) away is the bad version. Where the mid sits relative to the glass matters more than how far it is from the tweeter. The [Changed view] tag: in 2012 and March 2013 he advised keeping speakers, mids especially, away from that junction; from August 2013, and firmly in 2021, he said tight in the corner is best and hanging near it is the problem.

1.6 #9 Dash mounting gives big free boundary gain and turns the windshield into a "tweeter".

> Each halving of radiating space adds ~6 dB (dash, then windshield), so a 2.5" dash speaker plays down to 200–250 Hz on little power; cut the hump rather than boost the top. Aimed at a raked windshield, its beamed highs reflect to everyone: the factory "middler" is a "poor man's tweeter" (fails with upright glass, e.g. Jeep). RARE **Conflict:** see 1.6 #23 on firing mids into the glass.
>
> **Sources:** Audiofrog blog Jun 2021 · CMA Connected Oct 2021 @32:43 · DIYMA 2022-01-03

In plain English: Boundary gain: a speaker near a surface can only send sound into part of the space around it, so its lower notes get louder. He says each halving of that space adds about 6 dB: once for the dash, again for the windshield. That lets a 2.5" (about 65 mm) dash speaker play down to 200–250 Hz on little power. His advice is to cut the resulting hump with EQ rather than boost everything above it. And because the small speaker beams its highs, aiming it at a sloped ("raked") windshield bounces those highs to everyone. That is why he calls the factory dash "middler" (a small dash midrange) a "poor man's tweeter". It fails with upright glass, such as a Jeep's. The [Conflict] tag points to 1.6 #23, where he criticizes firing midranges into the glass.

1.6 #10 Dash-top left/right speakers multiply sources.

> Add a center and you get "three left speakers, three right speakers and two center channels" (direct plus reflections), which confuses auto time alignment (try reversing the center's polarity). Splitting a 2-way so mid and tweeter sit together on the dash is pointless: you mostly hear the windshield.
>
> **Sources:** DIYMA 2011-05-12 · DIYMA 2012-04-06

In plain English: Speakers on top of the dash bounce off the windshield, so each one is heard more than once: directly and again as reflections. He says that adding a center speaker there gives you "three left speakers, three right speakers and two center channels". All those copies confuse automatic time alignment (a processor routine that measures when each speaker's sound arrives and sets delays); his suggestion is to try reversing the center's polarity (swapping its + and − connections). He also says splitting a 2-way set (a midrange plus a tweeter) so that both sit together on the dash is pointless, because what you mostly hear is the windshield.

1.6 #11 Dash tweeters: far forward and outboard, or hidden under the factory grille.

> Tweeter type matters "FAR less" than position: put it near the dash–windshield–pillar junction, where aiming up or forward barely matters. Behind a factory grille, clear the plastic and hide a small tweeter ("No need for ugly tweeter pods"), avoiding a ~5/8" gap between its base and the glass (that spacing puts a dip near 10 kHz). **RARE**
>
> **Sources:** DIYMA 2013-04-25 · DIYMA 2021-09-05 · DIYMA 2021-09-04

In plain English: If your tweeters go on the dash, he says to put them far forward and toward the outside, near where the dash, windshield and pillar meet. There, whether you aim them up or forward barely matters, and which tweeter you buy matters "FAR less" than where you put it. If the dash has a factory speaker grille, he suggests clearing the plastic behind it and hiding a small tweeter underneath ("No need for ugly tweeter pods"). He warns against the tweeter's base sitting about 5/8" (16 mm) from the glass, because that spacing causes a dip (a missing band of sound) near 10 kHz.

1.6 #12 Kick panels fix path lengths but are "terrible for frequency response".

> They cut the left/right path difference from ~28" to ~6" and give real two-seat midrange imaging, but the footwell makes a big peak and dip around 300–800 Hz, legs block them and the stage narrows. They work only when the vertical angle to your ears is under ~20° (sedans, not trucks or minivans). **RARE**
>
> **Sources:** Time Alignment Part 2 · DIYMA MS-8 FAQ #13 · DIYMA 2016-04-27

In plain English: Kick panels are the side panels down in the footwells, in front of your feet. From either front seat, the far-side speaker is much farther away than the near-side one; he says kick panels cut that difference from about 28" (71 cm) to about 6" (15 cm). That gives real midrange imaging (sounds that seem to sit in the right places) for both front seats. But he calls them "terrible for frequency response": the footwell causes a big peak and dip somewhere around 300–800 Hz, your legs block the sound, and the stage gets narrower. He says they only work when the angle from the speaker up to your ears is under about 20°, which fits sedans but not trucks or minivans.

1.6 #13 Doors plus a center beat kick panels for most people.

> Kicks create about as many problems as they solve; with a center, doors give a wider stage; for a one-seat car with delay, "don't bother with the kicks." With no delay, little EQ and no center, pick kicks or any layout keeping the left/right path difference under ~1 ft. RARE
>
> **Sources:** DIYMA MS-8 FAQ #13 · DIYMA MS-8 FAQ p4 #71 · DIYMA 2021-06-29

In plain English: For most people, he says, doors plus a center speaker beat kick panels. Kicks create about as many problems as they solve. With a center channel (a speaker in the middle of the dash), doors give a wider stage. In a car tuned for one seat with delay (time alignment) available, "don't bother with the kicks." The exception: with no delay, little EQ and no center, pick kicks, or any layout that keeps the difference between your distances to the left and right speakers under about 1 ft (30 cm). Two of the sources come from his FAQ for the JBL MS-8 processor, so they date from that product's era.

1.6 #14 Every location has its own reflection dip, so choose the trade-off instead of chasing a perfect spot.

> His own low-door mids have a side-glass dip (~300–500 Hz), the price of width; all-in-the-dash fixes that but loses width and adds a windshield dip; kicks add a midbass dip. "It's always something." RARE
>
> **Sources:** DIYMA 2021-11-02

In plain English: Every mounting spot has some nearby surface whose reflection cancels part of the sound and leaves a dip at some frequency. He gives examples. In his own cars, the low-door speakers get a dip from the side window around 300–500 Hz; that is the price of a wide stage. Putting everything in the dash fixes that dip but loses width and adds a dip from the windshield. Kick panels add a dip in the midbass. His summary: "It's always something." So in practice: pick the trade-off you can live with rather than hunting for a perfect location.

1.6 #15 The late-model BMW layout is close to "optimum".

> 4" door mid, sail tweeter, 4" dash center and under-seat woofers: a good directivity match, central woofers that don't overload rear passengers, and seats that soak up woofer noise. His MS-8 BMW crossed the under-seat woofers (~80–200 Hz) at 200 Hz, as low as the 4" doors could play loudly; keep an octave of usable bandwidth around any crossover. RARE
>
> **Sources:** DIYMA 2013-03-10 · DIYMA MS-8 FAQ #12 · DIYMA 2021-08-04

In plain English: He calls the layout in late-model BMWs close to "optimum": a 4" (about 100 mm) midrange in each door, a tweeter in each sail panel, a 4" center on the dash, and woofers under the front seats. His reasons: the speakers' directivity (how widely each spreads its sound) matches well

from one to the next; woofers near the middle of the car don't overload rear passengers; and the seats soak up woofer noise. In the BMW he set up with the MS-8 processor, the under-seat woofers played roughly 80–200 Hz and were crossed at 200 Hz, as low as the 4" door speakers could play loudly. His general rule: keep about an octave (a halving or doubling of frequency) of usable range around any crossover, so the speakers on both sides of it still play well past the hand-over point.

1.6 #16 Multiple midbass locations even out the seats but wreck imaging above ~100 Hz.

> He tried 4, 6 and 8 midbass drivers with delays, levels and EQ: the more even the seats, the worse the image, and it "doesn't really work very well" for the cost. Keep everything below ~100 Hz out of the rears unless it cancels the sub for the back seat (why BMW under-seat woofers and Mercedes "Front Bass" work). RARE
>
> **Sources:** DIYMA 2009-11-11 · DIYMA 2021-11-29

In plain English: Some people spread several midbass speakers around the car so every seat gets even bass. He tried 4, 6 and 8 midbass drivers, adjusting delays, levels and EQ for each. The more even the seats became, the worse the imaging got above about 100 Hz, and it "doesn't really work very well", especially for the cost. His rule: keep everything below about 100 Hz out of the rear speakers, unless that sound is used to cancel some of the subwoofer for the back seat, which is often much closer to the sub. That, he says, is why BMW's under-seat woofers and Mercedes' "Front Bass" system work.

1.6 #17 Behind the rear seat is the worst subwoofer location, though the easiest to design for.

> A deep null at the seat can come from where sub and listener sit, not crossover phase; a dip with only one of two subs is likely the cross-car mode (direct and far-side-bounce paths differing by ~7 ft). RARE
>
> **Sources:** DIYMA 2021-09-16 · DIYMA 2021-12-02

In plain English: In many cars the subwoofer goes in the trunk, behind the rear seat. He calls that the worst place for it, though the easiest to design for. If you measure a deep null (a very deep dip at one frequency) at your seat, he says it can come from where the sub and the listener sit, not from crossover phase (the sub and midbass being out of step). If the dip appears with only one of two subs, it is likely the cross-car mode. He describes it by its paths: the sound going straight from the sub to you and the sound bouncing off the far side of the car travel distances that differ by about 7 ft (2.1 m).

1.6 #18 Home driver-spacing rules matter less in cars, with two exceptions.

> Cars have no flat baffles and use steep active slopes, and mid and midbass sharing a door's air is "no big deal". But stack the tweeter directly above the mid (aligned at every horizontal angle), and never in front of a moving woofer cone, whose motion causes Doppler "warbling" (coaxials below ~8 kHz). RARE
>
> **Sources:** DIYMA 2009-06-19 · DIYMA 2021-09-07 · DIYMA 2013-11-29

In plain English: Home speaker builders follow rules about how close together the drivers must sit on the front panel (the baffle). He says those rules matter less in cars, which have no flat baffles and use steep active crossovers (electronic filters that cut off sharply). So a midrange and a

midbass sharing the same air inside a door is "no big deal". There are two exceptions. First, stack the tweeter directly above the midrange, so the two stay lined up from every side-to-side angle. Second, never put a tweeter in front of a moving woofer cone, as coaxial speakers do (the tweeter sits in the middle of the woofer). The moving cone causes a Doppler "warbling" in the tweeter's sound, which he says affects coaxials below about 8 kHz.

1.6 #19 Map candidate locations with a mic and a speaker before cutting anything.

> A sound path works the same in both directions (reciprocity), so put a small speaker at the driver's head and a boundary mic at each candidate spot; it works for sub, midbass and mids, not tweeters. By 2023 this "system design by mapping" was his MasterTech seminar, replacing guessed rules like "avoid the windshield corner". RARE
>
> **Sources:** OFCAP Ep. 38 @36:00 · DIYMA 2021-11-07 · Absolute Electronix 2023 @1:00:14

In plain English: Before cutting holes, he suggests measuring. Reciprocity is a basic acoustics fact: sound travels the same path in both directions, so a speaker at A heard by a microphone at B measures the same as a speaker at B heard at A. So put a small speaker where the driver's head would be and a boundary microphone (a flat mic made to lie on a surface) at each spot you are considering. Each measurement shows how a speaker in that spot would reach your head. He says this works for subwoofers, midbass and midranges, but not tweeters. By 2023 this "system design by mapping" was the subject of his MasterTech seminar, replacing guessed rules like "avoid the windshield corner".

1.6 #20 Use the factory locations when they work; experts chose them for a reason.

> OEM engineers (many with PhDs) place speakers deliberately, and rearranging a car to install "the thing we've always installed" is a big mistake. Fabricate only for a placement you need, such as a factory-looking mid at the top of the door; "Every installation is a compromise." RARE
>
> **Sources:** CMA Connected Oct 2021 @37:25 · CMA Connected Oct 2021 @41:52 · OFCAP Ep. 38 @32:04

In plain English: OEM means the carmaker's own factory engineering. He says these engineers, many with PhDs, place speakers deliberately, and that rearranging a car to install "the thing we've always installed" is a big mistake. So in practice: use the factory locations when they work, and only build custom mounts for a placement you actually need, such as a factory-looking midrange at the top of the door. His reminder: "Every installation is a compromise."

1.6 #21 No time alignment available? Choose locations that need less of it, and add a real center.

> Then the center image doesn't depend on a perfect acoustic sum; if the owner doesn't care about imaging, drivers can go anywhere. Expect car-specific exceptions: a big instrument-cluster bulge can spoil sail tweeters for the passenger.
>
> **Sources:** DIYMA 2014-06-11 · DIYMA 2011-05-12

In plain English: Time alignment is a processor feature that delays the closer speakers so all the sound reaches you together. If you don't have it, he says to choose locations that need less of it and to add a real center speaker. A real center plays the middle of the music itself, so the center image doesn't depend on the left and right sounds adding up perfectly at your ears. If the owner doesn't care about imaging, the speakers can go anywhere. He also warns to expect car-specific exceptions: a big bulge over the instrument cluster can spoil sail-panel tweeters for the passenger.

1.6 #22 Judge placements by criteria, not trophies.

> He had never seen a rebuilt dashboard shaped by anything measured; "Joe Bastardo won IASCA with speakers in the glove box" is not evidence, and a competition win doesn't create new physics. RARE
>
> **Sources:** DIYMA 2009-07-24 · DIYMA 2020-10-07

In plain English: He says to judge a speaker location by clear criteria, not by trophies. He had never seen a rebuilt dashboard shaped by anything that had been measured. A line like "Joe Bastardo won IASCA with speakers in the glove box" is not evidence (IASCA is a car-audio sound-quality competition), and a competition win doesn't create new physics. Many people believe that if a winning competition car used a layout, it must be right; this point disagrees.

1.6 #23 Firing midranges into the glass: he moved against it.

> In 2009 his own install fired into the glass and he EQ'd the hole (vocals sat right). From 2011 he said windshield-firing narrows the stage, and in 2021 that firing mids up at the glass or ceiling makes "the most dense comb filter possible". He keeps exceptions for an up-firing center and factory dash "middlers" (1.6 #9). **Changed view:** 2009 → 2011/2021 **Conflict:** he praises the windshield-aimed middler (Oct 2021–Jan 2022) a few months after condemning up-firing mids (Sep 2021).
>
> **Sources:** DIYMA 2009-07-10 · DIYMA 2011-03-26 · DIYMA 2021-09-07

In plain English: This point tracks how his view on aiming midranges at the windshield changed. In 2009 his own car fired its mids into the glass, and he used EQ to fill in the resulting hole (a dip in the sound); vocals sat in the right place. From 2011 he said firing at the windshield narrows the stage, and in 2021 that pointing mids up at the glass or the ceiling makes "the most dense comb filter possible" (the densest pattern of reflection dips). He still makes exceptions for a center speaker firing up at the windshield and for factory dash "middlers" (see 1.6 #9). The [Conflict] tag: in September 2021 he condemned up-firing mids, yet from October 2021 to January 2022 he praised the windshield-aimed middler. Both statements are his.

1.7 Stage width, height and depth

What this part is about: The "soundstage" is the imaginary stage you seem to hear in front of you. This part covers what decides how wide, how high and how deep that stage seems, and which popular tricks he says don't work.

1.7 #1 Stage width is fixed by where the speakers are: the inverted triangle.

> With you at the bottom point, the stage can't be wider than the speakers, and a sound in one speaker comes from that speaker (his football-field test). After delays and EQ, pan the balance fully left, then right: those are your real edges. Sail tweeters widen the triangle; A-pillars and windshield reflections narrow it. RARE
>
> **Sources:** DIYMA 2008-05-20 · DIYMA 2017-08-18 · DIYMA 2017-08-21

In plain English: Picture an upside-down triangle: you sit at the bottom point and the two speakers sit at the top corners. He says the stage can't be wider than the speakers. His football-field test shows why: put two speakers in the middle of a football field, where nothing reflects, turn the balance all the way left, and the sound comes from the left speaker. A sound that plays

from only one speaker comes from that speaker. So in practice: after setting delays and EQ, pan the balance fully left, then fully right; those two positions are your real stage edges. Sail-panel tweeters widen the triangle; A-pillars and windshield reflections narrow it.

1.7 #2 No two-channel trick reliably widens the stage; "width" from delay games is comb filtering.

> "Play with delays for width" is a myth: the out-of-phase bands just expose speaker locations (1.1 #3). Images can't be generated reliably outside the speaker–listener triangle; real extra width needs side or rear speakers and processing. With kick mids and door tweeters, more tweeter level adds a little.
>
> RARE
>
> **Sources:** DIYMA 2021-08-14 · DIYMA 2017-08-21 · DIYMA 2017-08-18

In plain English: Many people believe you can widen the stage by adjusting the delays between left and right. He calls "play with delays for width" a myth. Those settings create bands of frequencies where left and right are out of phase (working against each other), which is a form of comb filtering, and he says that just exposes where the speakers really are (see 1.1 #3). In his view, sounds can't be placed reliably outside the triangle formed by the speakers and the listener. Real extra width needs side or rear speakers plus processing. One small exception: with mids in the kick panels and tweeters in the doors, more tweeter level adds a little width.

1.7 #3 In a room, width comes from the side-wall reflection; in a car, crosstalk takes it away.

> At home the first side-wall reflection acts like a phantom speaker outside the wall, making the room seem bigger without moving instruments. In a car, reflections mark the car's boundary: side-glass bounces add a little width, windshield reflections narrow it, and pillar mids just make images "big".
>
> RARE
>
> **Sources:** DIYMA 2011-11-17 · DIYMA 2017-08-21

In plain English: At home, sound from each speaker bounces off the nearby side wall. He says that first side-wall reflection acts like a phantom speaker beyond the wall, much like a mirror image, so the room seems bigger without the instruments moving. A car works differently: its reflections mark the car's own boundary, and crosstalk (sound meant for one side spreading to the other) takes width away. Bounces off the side windows add a little width, windshield reflections narrow it, and midranges in the pillars just make images "big" (spread out) rather than wider.

1.7 #4 "Stage outside the car" claims rest on one or two demo tracks.

> Typical evidence is a single event on a judging disc; his own room-synthesis DSP put the EMMA disc's walking man 10 ft beyond each side, but not on every track. RARE
>
> **Sources:** DIYMA 2011-11-17

In plain English: Some people claim their system puts the stage outside the car. He says such claims usually rest on one or two demo tracks, often a single effect on a competition judging disc. His own room-synthesis processor (electronics that add artificial room reflections) placed the "walking man" on the EMMA test disc (EMMA is a car-audio competition) about 10 ft (3 m) beyond each side of the car, but that didn't happen on every track.

1.7 #5 Depth is not distance to the stage; it comes from ambience in the recording or added by processing.

> Judges confuse the two; sitting closer to the speaker line raises width and "depth" but softens focus. Real depth comes from early reflections recorded or synthesized in the track, rears with processing, or room synthesis; not every recording has it, so don't retune the car to find it. RARE
>
> **Sources:** DIYMA 2013-04-27 · OFCAP Ep. 19 @3:23:07 · DIYMA 2022-03-03

In plain English: "Depth" means sounds seeming to sit at different distances, some near and some farther back. He says depth is not the same as how far away the stage seems, and that judges confuse the two. Sitting closer to the line between the speakers makes the stage seem wider and "deeper" but less sharply focused. Real depth comes from ambience: early reflections captured in the recording or synthesized by processing, rear speakers fed by processing, or room synthesis. Not every recording has depth, so in practice: don't retune your car trying to find it.

1.7 #6 Aiming changes tone far more than width or depth.

> His home test: move your chair toward the speaker plane, re-aiming on-axis; a stage farther in front of you is a change of seat, not image depth. Doors can beat on-axis mounting for both. RARE
>
> **Sources:** DIYMA 2013-04-27

In plain English: He says pointing speakers changes their tone far more than it changes stage width or depth. His home test: move your chair forward until you are almost in line with the speakers, turning them so they keep pointing at you. In his account the depth survives, which suggests speakers beside you, as in the doors, can still give depth and width. A stage that sits farther in front of you is simply a different seat position, not real image depth. He adds that door mounting can beat on-axis (aimed-at-you) mounting for both width and depth.

1.7 #7 Stage height comes from tweeter location plus crossover phase, not EQ.

> Put tweeters high (dash, door tops, sail panels); then correct phase at each crossover pulls the whole range up to tweeter height, so with a proper crossover height stops being a problem and pillar mids are a quick fix, like subs up front. RARE **Changed view:** 2016 stressed physical location; 2020–21 adds that crossover phase holds everything at tweeter height.
>
> **Sources:** DIYMA 2016-04-27 · DIYMA 2020-10-07 · DIYMA 2021-09-07

In plain English: Stage height is how high the music seems to sit, for example at dash level rather than down by your knees. He says it comes from where the tweeters are plus correct crossover phase, not from EQ. Put the tweeters high: on the dash, at the tops of the doors or in the sail panels. Then, if each crossover hand-over is in phase (the two speakers sharing those notes move in step), the whole range is pulled up to tweeter height. With a proper crossover, he says, height stops being a problem, and he describes pillar mids as a quick fix, comparable to putting subwoofers up front. The [Changed view] tag: in 2016 he stressed physical location; in 2020–21 he added that crossover phase holds everything at tweeter height. Many people believe EQ can raise the stage; this point disagrees.

1.7 #8 For height plus width: mids at the top of the doors, tweeters in the sail panels, a center where the gauge pod was.

> An up-firing center reflects off the middle of the windshield, giving height and width at once ("Move the gauges to the a-pillar."). With center steering there's no phantom center to build, so L/R tweeters are free to lift and widen the stage. RARE
>
> **Sources:** DIYMA 2013-08-31 · DIYMA MS-8 FAQ p2

In plain English: For a stage that is both high and wide, he suggests midranges at the tops of the doors, tweeters in the sail panels, and a center speaker where the gauge pod (instrument cluster) sits. A center speaker firing up at the windshield reflects off the middle of the glass, giving height and width at once. As he put it, "Move the gauges to the a-pillar." With center steering (a processor that sends the middle of the music to the center speaker), the left and right speakers no longer have to create a phantom center (the illusion of a middle sound made by both sides playing it equally). That frees the left and right tweeters to lift and widen the stage. One source is his MS-8 FAQ, so part of this relates to that processor.

1.7 #9 "Stage base a third up the pillars, vocals slightly above" is one recording's layout, not a rule.

> It holds for a dry studio track with the singer panned center. Height-test tracks drove the Asian fad for pillar pods with 6" drivers in boxes barely bigger than the basket: "a super high-Q midrange that doesn't play any bass". RARE
>
> **Sources:** DIYMA 2013-04-29 · DIYMA MS-8 FAQ p5 #95

In plain English: The quoted rule says the base of the stage should sit a third of the way up the A-pillars, with vocals slightly above. He says that is just how one kind of recording is laid out, not a rule: it holds for a dry studio track (little room echo) with the singer panned to the center. He adds that height-test tracks drove a fad in Asia for pillar pods holding 6" (about 150 mm) speakers in boxes barely bigger than the speaker's frame (its basket), giving "a super high-Q midrange that doesn't play any bass" (high-Q meaning a peaky response).

1.8 Tweeterless and widebander systems

What this part is about: Some builders skip tweeters and let one small "full-range" or "widebander" speaker play both the midrange and the highs. He argues firmly for using a tweeter, and explains what goes wrong without one and how his own small midranges were designed.

1.8 #1 Always use a tweeter; there's no credible case for skipping it in a car.

> Tweeters are the smallest, cheapest, easiest drivers to fit. A car is near-field listening with road noise (so crossover insertion loss is irrelevant) and reflectors close enough that you hear the power response; a 3" run from 100 Hz to 20 kHz is inefficient and distorts. In 2021 he still said every car sounds better with tweeters. RARE
>
> **Sources:** DIYMA 2014-04-13 · DIYMA 2015-05-07 · DIYMA 2021-12-02

In plain English: He says always use a tweeter; there's no credible case for skipping one in a car. Tweeters are the smallest, cheapest and easiest speakers to fit. In a car you sit close to the speakers (near-field listening) with road noise around you, so he says the small loss a crossover

adds (its "insertion loss") is irrelevant. Reflecting surfaces are also so close that you hear the power response, meaning all the sound a speaker sends out in every direction, not just straight ahead. A 3" (about 75 mm) speaker made to play from 100 Hz to 20 kHz is inefficient and distorts. In 2021 he still said every car sounds better with tweeters. Many people believe a single full-range speaker sounds purer than a midrange plus a tweeter; this point disagrees.

1.8 #2 Tweeterless sounds tiny, precise and lifeless.

> Images are small and very precise, but the sense of space carried by high-frequency reflections is lost (HF reflections please most listeners). It may score IASCA points; he prefers a believable space. A sail tweeter "won't screw up" full-range drivers; it improves them. RARE
>
> **Sources:** DIYMA 2014-10-30 · DIYMA 2013-05-27 · DIYMA 2013-05-23

In plain English: Without a tweeter, he says, images become small and very precise, but the sense of space is lost, because that space is carried by high-frequency reflections, which please most listeners. Such a system may score points at IASCA (a sound-quality competition), but he prefers a believable space. Adding a tweeter in the sail panel "won't screw up" full-range drivers; he says it improves them.

1.8 #3 Widebanders were designed for desktops, and car builders use them for the wrong job.

> They target a listener 18" away with walls 10 ft off, trading sensitivity for low resonance and stretching treble to save a tweeter; cars then pair them with a midbass, removing their reason to exist. No 3" is truly full-range if off-axis response matters; the no-tweeter pitch is "marketing hooey". RARE
>
> **Sources:** DIYMA 2013-09-18 · DIYMA 2015-08-23

In plain English: A widebander is a small speaker stretched to cover most of the audio range. He says they were designed for desktops: a listener about 18" (45 cm) away, with walls about 10 ft (3 m) off. They trade sensitivity (how loud they play per watt) for a low resonance (so they reach lower notes), and stretch their treble to save the cost of a tweeter. Car builders then pair them with a midbass, which removes their reason to exist. No 3" (about 75 mm) speaker is truly full-range if its sound off to the side (off-axis) matters, he says, and the no-tweeter sales pitch is "marketing hooey".

1.8 #4 If you run a widebander without a tweeter, aim it at the windshield.

> Aimed at you, highs arrive direct but the reflections carry few: tiny "clinical" images with no space. Aimed elsewhere, highs live in weak reflections: dull and lifeless. Aimed at raked glass from the dash (as OEMs do), the windshield becomes the HF source, "a tweeter without having to pay for a tweeter". A real tweeter still wins. RARE **Changed view:** 2012–14 he said a tweeterless widebander must be on-axis; 2018–21 he ranks windshield-aimed best.
>
> **Sources:** DIYMA 2012-09-06 · DIYMA 2018-03-31 · DIYMA 2021-12-02

In plain English: If you do run a widebander without a tweeter, he says to aim it at the windshield. Aimed at you, its highs arrive directly but the reflections carry few highs, so images are tiny and "clinical", with no sense of space. Aimed elsewhere, the highs exist only in weak reflections, so it sounds dull and lifeless. Mounted on the dash and aimed at a sloped windshield, as carmakers do,

the glass becomes the source of the highs: "a tweeter without having to pay for a tweeter". A real tweeter still wins. The [Changed view] tag: in 2012–14 he said a tweeterless widebander must be aimed at the listener (on-axis); in 2018–21 he ranked windshield-aimed best.

1.8 #5 Full-range pillar pods can't be EQ'd for two seats.

> Forward imaging needs similar directivity above at least 1 kHz (better, above 100 Hz). A small full-range pillar pod gives midrange spread from three nearby surfaces but pinpoint highs, so an EQ that works in one seat is "horrible" in the other. RARE
>
> **Sources:** DIYMA 2013-09-17

In plain English: Directivity describes how widely a speaker spreads its sound at each frequency. He says that for the stage to seem to sit in front of you, the directivity needs to stay similar across all frequencies above at least 1 kHz, and better above 100 Hz. A small full-range speaker in a pillar pod doesn't manage that: its midrange spreads out and bounces off three nearby surfaces, while its highs come from one pinpoint spot. As a result, an EQ that works in one seat is "horrible" in the other.

1.8 #6 Audiofrog's small mids are midranges, not widebanders.

> The GB25 brief: reach 200 Hz in a 0.004 cu ft box at 100 W RMS, efficiency matched to its midbass, and a top-end peak the low-pass removes so a tweeter takes over. Going tweeterless means EQing that peak out ("please, God no!").
>
> **Sources:** DIYMA 2015-05-07 · DIYMA 2016-03-03

In plain English: Audiofrog's small speakers, like the 2.5" (about 65 mm) GB25, are midranges meant to work with a tweeter, not widebanders. The GB25's design brief: play down to 200 Hz in a tiny 0.004 cu ft box (about 0.11 litres, less than half a cup) at 100 W RMS (continuous power); match the efficiency (loudness per watt) of its midbass partner; and have a peak at the top of its range that the low-pass filter (which cuts the notes above the crossover) removes, so a tweeter takes over. Running it without a tweeter would mean using EQ to flatten that peak, to which he says "please, God no!".

1.9 One seat versus two seats

What this part is about: In a car nobody sits in the middle, so a tune that is perfect for the driver can sound lopsided for the passenger. This part covers his recipes for one seat versus two seats, why time delay helps one seat and hurts the other, and the tools he uses instead: all-pass filters (which shift timing without changing loudness) and a center speaker.

1.9 #1 His ranked recipes: one seat, two seats, and "go nuts".

> One seat: a 3-way in each door plus delay. Two seats: the same doors with sail tweeters, a big center and processing that extracts the center (the whole point of a center is that both front seats hear stereo). One seat, "go nuts": add a center, crosstalk cancellation and synthesized reflections. RARE
>
> **Sources:** DIYMA MS-8 FAQ p5 #98 · DIYMA 2013-08-02 · DIYMA 2022-02-24

In plain English: A "3-way" splits each side's sound across three speakers: a woofer for low notes (midbass), a midrange for the middle, and a tweeter for high notes. "Delay" (time alignment) means a processor holds back the nearer speakers so all the sound reaches your ears at the same moment. Sail panels are the small triangular panels at the front corner of each door, by the mirror. A center speaker sits in the middle of the dash. "Extracting the center" means a processor pulls out only the sound that left and right share and sends it there. He ranks three recipes. For one listener: a 3-way in each door plus delay. For two front seats: the same doors with sail tweeters, a big center and center-extracting processing, because the whole point of a center is that both seats hear stereo. For one seat, going all out ("go nuts"): add a center, crosstalk cancellation (processing that stops each ear from hearing the opposite speaker) and synthesized (artificially created) reflections.

1.9 #2 Delay fixes one seat and doubles the error in the other.

> The distance error is equal and opposite for the two front seats. His rule: "the passenger's seat can't suck"; a tuner who leaves other seats sounding like garbage is "a learner", not a "tuning God". Carmakers back the two-seat goal with heavy market research. RARE
>
> **Sources:** DIYMA 2016-03-16 · D&F live @26:03 · DIYMA 2021-09-09

In plain English: You always sit closer to one front speaker than the other, so its sound arrives first. Delay holds back the near speaker so both arrive together. The catch: the passenger sits in the mirror-image spot. The speaker that is "near" for the driver is "far" for the passenger, so delaying it makes it arrive even later there. The passenger's timing error was already the same size as the driver's, just the other way round ("equal and opposite"); after the delay it is twice as big. His rule is that "the passenger's seat can't suck". A tuner who leaves the other seats sounding like garbage is, in his words, "a learner", not a "tuning God". He adds that carmakers back the two-seat goal with heavy market research. Many people believe a tune is finished once the driver's seat sounds perfect; he disagrees.

1.9 #3 A-pillar cars without a center are always one-seat cars.

> He has never heard one that wasn't; centered vocals need no center speaker only in a car optimized for one seat. RARE
>
> **Sources:** DIYMA 2014-06-16 · DIYMA 2021-12-07

In plain English: A-pillars are the posts on each side of the windshield, a popular spot for speakers. Many people believe that speakers up there, well tuned, give everyone a centered singer without a center speaker. He says he has never heard an A-pillar car without a center that wasn't a one-seat car. A singer can sound centered with no center speaker only when the car is optimized for one seat; move to the other seat and the center falls apart.

1.9 #4 What breaks the image for both seats is phase near 250 Hz, and an all-pass fixes it without delay.

> With door speakers, left and right sit near −180° at ~250 Hz in both seats; an all-pass reverses one speaker's phase over a narrow band there, helping both seats at once: not perfect in either, but better. RARE **Changed view:** 2013–15 he judged phase EQ inaudible next to delay for one seat; in Jan 2016 he said new DSPs

> should drop L/R delay for phase EQ.
>
> **Sources:** D&F live @21:00 · DIYMA 2016-01-25 · DIYMA 2016-03-16

In plain English: Sound is a wave, and "phase" describes where in its up-and-down cycle the wave is at a given moment. If two speakers play the same note but one is pushing while the other is pulling (180 degrees apart), they cancel, like two people pushing a swing from opposite sides. Because of the different distances to the door speakers, left and right arrive about half a cycle apart near 250 Hz (a lower-midrange note) in both front seats. Half a cycle early or half a cycle late cancels the same way, which is why both seats suffer. An all-pass filter is a processing tool that leaves loudness alone and only shifts the phase of a chosen band. He says putting one on one speaker reverses its phase over a narrow band there, which helps both seats at once: not perfect in either, but better. Delay cannot do that. His view changed: in 2013–15 he judged phase EQ (adjusting phase by frequency) inaudible next to delay for one seat; in January 2016 he said new processors should drop left/right delay in favor of phase EQ.

1.9 #5 Carmakers already do it: Toyota, Subaru, Bose and a Harman patent.

> Toyota uses all-pass filters from basic systems up to JBL; Bose's Fiat 500 images in both seats with plain stereo. In such cars add no front delay and leave "reversed" factory polarity alone: the filters can't be inverted, and added delay makes them work poorly. RARE **Conflict:** he recalls Harman patent US6876748 naming 250 Hz; its text cites about 200 Hz.
>
> **Sources:** D&F live @23:05 · DIYMA 2021-05-13 · US6876748 (Google Patents)

In plain English: He says carmakers already use this all-pass approach, naming Toyota, Subaru and Bose. Toyota uses all-pass filters from its basic systems up to its JBL premium systems, and Bose's Fiat 500 system makes a good image in both seats with plain stereo (no center). A Harman patent covers the idea. Polarity is which way a speaker's cone moves for a given signal; a speaker wired "reversed" moves in when it should move out. If you upgrade such a car, he says: add no front delay, and leave factory polarity that looks "reversed" alone. The filters can't be inverted (undone), and added delay makes them work poorly. The [Conflict] tag: he remembers the patent naming 250 Hz, but the patent's own text cites about 200 Hz. Both figures are given here; this explanation doesn't choose between them.

1.9 #6 Two quick tests show whether a factory system uses all-pass filters or plain delay.

> From each front seat, listen to a centered vocal: with all-pass it stays roughly central for the driver and both sides stay audible from the passenger seat; with delay it collapses into the passenger door. Or feed mono pink noise and view L+R: one big dip at 200–500 Hz means all-pass; an even comb means delay. RARE
>
> **Sources:** DIYMA 2021-10-26

In plain English: These tests tell you how a factory system handles the two seats. Test one: play a song with a singer mixed in the center and listen from each front seat. With all-pass filters, the singer stays roughly central (not pinpoint) from the driver's seat, and from the passenger seat you can still hear both sides. With plain delay, everything collapses into the passenger door. Test two: play mono pink noise (a steady hiss containing all frequencies, identical in both channels) and view left plus right combined (L+R) on your processor's input analyzer. One big dip somewhere between

200 and 500 Hz means all-pass. An even "comb" (a row of regularly spaced dips, like the teeth of a comb) means delay. That is because a delay shifts timing at every frequency, so cancellations repeat up the range, while an all-pass shifts only one region.

1.9 #7 His 2016 starting recipe: one 2nd-order all-pass near 250 Hz, Q about 1, on one side.

> Pick where half a wavelength equals the path difference; it needn't be exact. Size it with his phase "viewer": left/right phase differences of 130° or more count as 180°, and Q ≈ center frequency ÷ the region's width. Use phase EQ only from ~80 Hz to 1 kHz. RARE
>
> **Sources:** DIYMA 2016-03-16/18 · DIYMA 2016-03-17/04-03

In plain English: This is his 2016 recipe for setting up that all-pass filter. A "2nd-order" all-pass rotates the phase through a full cycle across a band of frequencies, reaching a half-cycle (180 degrees) at its center frequency. "Q" sets how wide or narrow the filter's effect is: a higher Q is narrower, and a Q of about 1 is fairly broad. Put it on one side only, near 250 Hz, at the frequency where half a wavelength equals the path difference (the difference between your distances to the left and right speakers), because that is where the two sides arrive exactly half a cycle apart. He says it needn't be exact. To size it, he used a phase "viewer" he built: subtract one side's measured phase from the other's, count any difference of 130 degrees or more as 180 (a cancellation zone), then set Q to roughly the center frequency divided by the zone's width. For example, a zone centered on 250 Hz and 250 Hz wide gives a Q of about 1. Use phase EQ only from about 80 Hz to 1 kHz.

1.9 #8 The FIR version: phase correction below 1 kHz only, and no left/right delay.

> FIR is a filter type that can shape phase freely. Stitch a single-mic measurement below 1 kHz to a spatial average above, leaving the top minimum-phase; with no L/R delay this may give a two-seat stereo car without a center ("watch 250 Hz"). Spending 64k taps on 10–20 kHz is excessive. RARE
>
> **Sources:** DIYMA 2016-03-16

In plain English: FIR (finite impulse response) is a kind of digital filter that can shape phase freely, separately from loudness. Its "taps" are the number of calculation steps; more taps means finer control and more processing. His recipe: correct phase only below 1 kHz. Measure below 1 kHz with the mic at a single spot, measure above 1 kHz as a spatial average (several mic spots averaged), and stitch the two together. Above 1 kHz, leave the phase "minimum-phase", meaning no deliberate phase correction there. Set no left/right delay. Done this way, he says, it may give a two-seat stereo car without a center speaker, with the caution to "watch 250 Hz". In the same 2016 thread he explains the 1 kHz split: above it, phase matters little, and an averaged measurement better represents what we hear. Separately, he notes that a 128k-tap filter puts about 64,000 (64k) of its taps between 10 and 20 kHz, the top octave. He calls that excessive, because most of the resolution lands there at the cost of processing power and latency (processing delay).

1.9 #9 Sub-to-midbass phase differs by seat even with a center: fix it with all-pass, not delay.

> In his car the sub-to-near-midbass distance was half a wavelength at the crossover, and delaying the near midbass pulled the bass toward that side for both seats. Fix: an all-pass on the sub plus one half an octave higher on one midbass, so no pair sits at −180°. RARE
>
> **Sources:** DIYMA 2016-01-22 · DIYMA 2016-01-28

In plain English: A subwoofer plays the deepest bass; the midbass woofers (here, in the doors) play the next range up. The crossover is the frequency where one hands over to the other. His point is that adding a center speaker does not fix this hand-over, because sub-to-midbass phase still differs between the seats. In his own car, the paths from the sub and from the nearer door midbass to the listener differed by half a wavelength at the crossover frequency, so the two cancelled there. Flipping polarity did not help, and delaying the near midbass pulled the bass toward that side in both seats. His fix: an all-pass filter on the sub, plus another set half an octave higher on one midbass (an octave is a doubling of frequency). That way no pair of these speakers sits at −180 degrees (cancelling), and he reports that the bass then sat centered in front of the dash in both front seats.

1.9 #10 All-pass imaging sounds bigger and less pinpoint, and by 2021 he wouldn't teach it.

> Driver's seat: a real but larger center than a delay-tuned "diorama of tiny army men on the dash"; passenger seat: more diffuse, still spread across the dash. By 2021 he called it fiddly (an all-pass on the later channel can create a new dip; try the other side) and teaches only repeatable processes.
>
> RARE **Changed view:** 2016 enthusiasm → 2021 "helps two seats, but only with fiddling".
>
> **Sources:** D&F live @23:05 · DIYMA 2021-09-07 · DIYMA 2021-08-27

In plain English: He compares the two approaches by ear. A delay tune for one seat gives a small, precise picture, which he likens to a "diorama of tiny army men on the dash" (a diorama is a miniature model scene). With all-pass filters, the driver's seat gets a real but larger, less pinpoint center; the passenger seat gets a more diffuse sound that still spreads across the dash. The [Changed view] tag: in 2016 he was enthusiastic, but by 2021 he called the method fiddly. An all-pass on the later-arriving channel can create a new dip, and then you try the other side. Because he teaches only repeatable processes, he would no longer teach it, though he still says it helps two seats with fiddling.

1.9 #11 MS-8 seat modes: Driver, Passenger and "Front".

> Driver or Passenger sets time alignment for that seat; "Front" drops L/R alignment but keeps it within each side. Without a center, Front balances tone but won't image; with one, the center gives good imaging in all seats. Gary Biggs won IASCA with stock BMW speakers, "Front" mode and the center up "a notch or two". **Era:** MS-8, 2008–2012
>
> **Sources:** DIYMA MS-8 FAQ #7 · DIYMA MS-8 FAQ #6 · DIYMA MS-8 FAQ p2

In plain English: The MS-8 was a JBL car processor sold around 2008–2012 that tuned a car automatically from microphone measurements; this point is about its settings. "Driver" or "Passenger" mode sets time alignment (delay) for that seat. "Front" mode drops the left/right alignment but keeps alignment within each side, for example between the tweeter and woofer on the same side. Without a center speaker, Front mode gives an even tonal balance but no focused image. With a center, the center provides good imaging in all seats. His example: Gary Biggs won IASCA (a car-audio sound-quality competition) with the stock BMW speakers, "Front" mode, and the center turned up "a notch or two".

1.9 #12 The polarity-flip trick is a pre-DSP half-fix; never combine it with delay.

> Reversing one mid fixes the ~250 Hz center cancellation but moves the problem to ~500 Hz and kills midbass; it dates from when polarity was the only tool. RARE
>
> **Sources:** Time Alignment Part 2 · DIYMA MS-8 FAQ #26 · D&F live @33:05

In plain English: Reversing polarity means swapping a speaker's plus and minus wires, so its cone moves the opposite way. An old trick was to reverse one front midrange to cure the cancellation near 250 Hz that ruins the center image. He calls it a half-fix from before processors existed, when polarity was the only tool. It moves the problem to about 500 Hz, where the two sides had been back in step, and it "kills" the midbass, because at low notes the two sides were in step and now cancel. His rule: never combine the flip with delay.

1.9 #13 Blur the center on purpose for listeners who hate it, without losing coherent bass.

> Ask in the parking lot where they hear the singer. In an active 3-way, delay the midbass but not mids and highs; in a 2-way plus sub, back delay off until drivers are in phase only around the ~100 Hz crossover. Never just switch delays off ("nobody hates coherent bass"); club-style listeners need a DSP for midbass-to-sub impact too. RARE
>
> **Sources:** OFCAP Ep. 38 @57:06 · OFCAP Ep. 38 @1:00:03 · CMA Connected Oct 2021 @46:57

In plain English: Some listeners don't like a sharply focused center image. He suggests asking them in the parking lot where they hear the singer. To blur the center on purpose without losing tight, unified bass: in an active 3-way (each speaker on its own amplifier channel, with a processor doing the crossovers), delay the midbass but not the mids and highs. In a 2-way plus sub, reduce the delay settings until the speakers are in phase only around the ~100 Hz crossover. His explanation: in phase, you hear an image between the speakers; out of phase, you hear the speakers themselves. Never just switch the delays off, because "nobody hates coherent bass". He adds that club-style listeners need a processor too, so the punch from midbass to sub still lines up.

1.9 #14 Judged cars game imaging; customers deserve "close to correct".

> Competitors tune per track: without per-track EQ he'd skip delay and use an all-pass for the center-vocal track, leaving everything above the vocal fundamental out of phase, which judges hear as width. Score sheets should reward two seats; until a car sounds larger than itself and alike in all seats, nobody should score a 10. RARE
>
> **Sources:** OFCAP Ep. 38 @54:01 · SQOLOGY P2 @21:02

In plain English: In judged competitions, he says, competitors tune for the specific test tracks. If he couldn't change EQ per track, he would skip delay and use an all-pass filter aimed at the center-vocal track, fixing phase around the singer's fundamental (the lowest main note of the voice). Everything above that would stay out of phase, which lets you hear where the speakers are, and judges hear that as width. His point is that judged cars game imaging this way. He thinks score sheets should reward cars that work in two seats, and that nobody should score a 10 until a car sounds larger than itself and alike in all seats. Winning a contest, he says, isn't the same as great audio; for customers, the goal is to get "close to correct".

1.10 The center channel

What this part is about: A center channel is a speaker in the middle of the dashboard. This part covers what signal a proper center should get, what kind of speaker it needs and where it goes, and how having one changes the way the rest of the system is tuned.

1.10 #1 "L+R is not a center": a summed mono center hurts more than it helps.

> There's "no happy medium" between loud enough to form a center and quiet enough to keep width, and passive L+R/L−R matrices "do not work". For a summed center: "turn it off and take the speaker out". Stereo music needs an upmixer, not a Dolby decoder. RARE
>
> **Sources:** DIYMA MS-8 FAQ #10 · DIYMA MS-8 FAQ p5 #82 · DIYMA 2018-04-06

In plain English: "L+R" means simply mixing the left and right channels together into one mono signal and sending that to a center speaker. Many people believe any center speaker fed this way helps. He says it hurts more than it helps. There is "no happy medium": loud enough to create a center image, it squeezes the stage narrow; quiet enough to keep the width, it doesn't create a center. Passive L+R/L−R matrices (simple wiring networks with no processor) "do not work" either. His advice for a summed center: "turn it off and take the speaker out". For stereo music you need an upmixer (a processor that splits stereo across more speakers), not a Dolby decoder (built for movie soundtracks).

1.10 #2 A real center gets only what L and R share, and that content stays in L and R about 6 dB down.

> A "hard center" of (L+R)−(L−R) is just 2R; extraction needs absolute values, which analog can't do. Off-center images form between L+C and C+R, so any polarity error spoils them. Remove center content from L/R entirely and images shrink; don't attenuate it and the stage narrows. RARE
>
> **Sources:** DIYMA MS-8 FAQ #11 · DIYMA 2017-07-01 · DIYMA 2018-03-29

In plain English: In a stereo recording, a sound "in the center", such as the lead singer, is identical in left and right. A proper center channel gets only that shared content, and the same content stays in left and right about 6 dB quieter (a clearly audible step down). He explains why simple math can't do this: a "hard center" of (L+R)−(L−R) works out to just 2R, a doubled right channel. True extraction needs absolute values (the size of a difference, ignoring its sign), which analog circuits can't calculate. Sounds slightly off-center, like a guitar just left of the singer, form between the left and center speakers or between the center and right ones, so any polarity error (a speaker wired backwards) spoils them. Getting the amount right matters: remove the center content from left and right entirely and images shrink; leave it at full level and the stage narrows.

1.10 #3 With real center processing, the center becomes the main front speaker.

> Most popular music is mixed center, so it must play as loud as L/R with similar response, include a tweeter, and cover 200 Hz–20 kHz at minimum (100 Hz is better), on main-channel power. Tweeters at the mirror are not a center. RARE
>
> **Sources:** DIYMA MS-8 FAQ p2 · DIYMA MS-8 FAQ p2 (tweeters) · DIYMA 2018-03-28

In plain English: Once a processor sends true center content to a center speaker, he says, that speaker becomes the main front speaker, because most popular music is mixed to the center. So it must play as loud as left and right, with a similar frequency response (similar tonal balance from low to high), and include a tweeter. It must cover 200 Hz to 20 kHz at a minimum, and reaching down to 100 Hz is better. It should run on main-channel power, the same kind of amplifier power the left and right get. In his view, tweeters at the mirror are not a center.

1.10 #4 Bass management follows the crossover you SET, so set the center's high-pass where it really plays.

> Content between what the center can reproduce and the high-pass you entered "is lost". Start with center high-pass 200 Hz and doors-to-sub 80 Hz; below the center's range the image depends on L/R time alignment again, so a 3" center with 8" doors means "there's gonna be trouble". RARE Era: MS-8
>
> **Sources:** DIYMA MS-8 FAQ p2 · DIYMA MS-8 FAQ p5 #93

In plain English: "Bass management" means the processor sends the low notes a speaker can't handle to other speakers: the center's lows to the left and right doors, and the doors' lows to the sub. A high-pass filter lets through only the notes above a set point. His warning is that the processor routes sound by the crossover numbers you type in, not by what the speaker can actually play. Content between the center's real lower limit and the high-pass you entered "is lost". His starting values: center high-pass at 200 Hz, doors-to-sub at 80 Hz. Below the center's range, the content plays from the left and right doors again, so the image depends on left/right time alignment again. That is why a 3-inch (about 75 mm) center with 8-inch (about 200 mm) door speakers means "there's gonna be trouble". This was MS-8-era advice.

1.10 #5 His preferred center: the biggest mid that fits plus a tweeter, on the dash near the windshield junction, firing up.

> Below ~1 kHz an 8" radiates evenly, so it's effectively on-axis whichever way it points, and the image sits between speaker and glass: higher, with "a nice, big center image". Seal its baffle. Its dip near 2 kHz is a feature, because ear spacing puts a similar dip in a phantom center (Shively and House, Harman AES papers, late 1990s). RARE
>
> **Sources:** DIYMA MS-8 FAQ p4 #71 · DIYMA MS-8 FAQ p5 #100 · DIYMA 2021-09-22

In plain English: His preferred center is the biggest midrange speaker that fits, plus a tweeter, mounted on the dash close to where it meets the windshield, firing upward. Below about 1 kHz, an 8-inch (about 200 mm) speaker spreads sound evenly in all directions, so it acts as though it is pointed at you whichever way it faces. The image then forms between the speaker and the glass, which places it higher and gives "a nice, big center image". Seal its baffle (the panel it is mounted in) so no air leaks around it. Such a speaker shows a dip near 2 kHz, which, he explains, comes from the reflection off the windshield. He says the dip is a feature, not a flaw: because of the spacing of our ears, a phantom center (the illusion of a center made by left and right speakers) has a similar dip. He credits this to late-1990s Harman AES papers by Shively and House.

1.10 #6 A center's line of sight barely matters.

> Up-firing and on-axis centers differ only above ~6 kHz without a tweeter (~10 kHz with one), like

> hearing someone yell from the kitchen.
>
> **Sources:** DIYMA 2013-12-05

In plain English: "On-axis" means the speaker points straight at you. He says it barely matters whether a center fires up at the windshield or points at you. The two sound different only above about 6 kHz (a high treble note) without a tweeter, or about 10 kHz with one. His comparison is hearing someone yell from the kitchen: you can't see them, but you hear them fine.

1.10 #7 No timbre matching needed: EQ matches the center to L/R.

> "No timbre matching by identical speakers is required"; EQ, not identical drivers, matches L/C/R. A coaxial works there, even a 2-way with its midbass in a small box behind the console. RARE
>
> **Sources:** DIYMA 2010-04-01 · DIYMA 2021-01-02 · DIYMA 2021-09-22

In plain English: Timbre is the tonal character of a sound. Many people believe the center must use the same model of speaker as the left and right so they sound alike. He says "No timbre matching by identical speakers is required": equalization (EQ, adjusting the loudness of different frequencies) is what makes left, center and right match. So a coaxial (a speaker with a tweeter mounted in its middle) works as a center, and so does a 2-way whose midbass sits in a small box behind the console.

1.10 #8 He has "NEVER heard" a good center in a Harman factory system.

> The factory center he replaced in 2016 made the whole car sound bad except the midbass, and he said a new algorithm's center target should have a deeper dip near 2 kHz. RARE
>
> **Sources:** DIYMA 2016-01-18 · DIYMA 2016-01-25

In plain English: He worked at Harman, which makes factory sound systems for many carmakers, yet he says he has "NEVER heard" a good center in a Harman factory system. The factory center he replaced in 2016 made the whole car sound bad except the midbass. He also said a new automatic-tuning algorithm should aim the center at a target with a deeper dip near 2 kHz (compare 1.10 #5).

1.10 #9 With a real center, L/R delay barely matters; place images with level.

> Steering exists "to minimize the importance of PLDs" (path-length differences). Give the center little or no delay (it's nearly equidistant), match its response to L/R, then aim for a mid-dash center for every seat: turn it down a bit to pull the image toward each listener, or up "a notch or two" for two seats. RARE
>
> **Sources:** DIYMA MS-8 FAQ #17 · DIYMA 2010-04-04 · EDUCAR

In plain English: "Steering" is when a processor moves each sound to the speaker where it belongs. He says it exists "to minimize the importance of PLDs", the path-length differences (your unequal distances to the left and right speakers) that delay normally has to fix. So with a real center, left/right delay barely matters, and you place images with level instead. The center needs little or no delay because it is nearly the same distance from both front seats. Match its response to left and right, then aim for a center image in the middle of the dash for every seat. Turning the center down a bit pulls the image toward each listener (some factory designers want it straight in front of each person); turning it up "a notch or two" suits two seats.

1.10 #10 Center polarity trick for windshield centers.

> With an up-firing center and L/R mids low in the doors or kicks, reversing the center's polarity and recalibrating sometimes focuses the center; first confirm the L/R mids are wired correctly.
>
> **Sources:** DIYMA 2013-03-05

In plain English: This is a trick for an up-firing center that bounces off the windshield while the left and right mids sit low, in the doors or in the kick panels (by your feet). Reversing the center speaker's polarity (swapping its plus and minus wires) and then recalibrating (re-running the automatic tune) sometimes makes the center image more focused. He says to first confirm that the left and right mids are wired correctly. He says "sometimes", so it is something to try, not a guaranteed fix.

1.11 Rear and side speakers

What this part is about: Many cars have speakers in the rear doors, on the rear shelf or in the back pillars. This part explains what he thinks those speakers are for (a sense of space, not sounds from behind you) and how to feed, place and set them, both with and without an upmixer.

1.11 #1 Rears are for envelopment, not rear events.

> The cabin's early reflections tell us we're in a car; later sound from the sides and rears lets us believe in a bigger space. With an upmixer, rears put you in the room instead of watching it through a window; a car can't sound like the Rose Bowl but can suggest a space larger than itself. RARE
>
> **Sources:** DIYMA MS-8 FAQ p3 · OFCAP Ep. 19 @3:28:03 · EDUCAR

In plain English: Many people believe rear speakers exist so you can hear sounds coming from behind, or that a serious sound-quality car should switch them off. He says rears are for envelopment, the feeling of being surrounded by the space of the recording. His reasoning: the early reflections in a car (sound bouncing off nearby surfaces, arriving just after the direct sound) tell your brain you are in a small space, a car. Sound that arrives later from the sides and rear lets you believe in a bigger space. With an upmixer, rears put you in the room instead of watching it through a window. A car can't sound like the Rose Bowl (a huge stadium), but it can suggest a space larger than itself.

1.11 #2 Rears are where extra width starts, if used carefully.

> Older recordings put out-of-phase "width" content in the opposite channel, which Logic7/PLII steer to the rears; rears at the right delay and level form phantom images between front and rear on each side. Rolling off their highs makes them harder to localize, and they sit far from the windshield. RARE
>
> **Sources:** DIYMA 2011-11-17

In plain English: He says rears are where extra width begins, if handled carefully. Older recordings often created "width" by placing a sound, phase-reversed, in the opposite channel. Matrix upmixers such as Logic7 and PLII (Pro Logic II) detect that out-of-phase content and steer it to the rears. Set at the right delay and level, the rears then form phantom images between the front and rear speakers on each side, so sounds can appear beside you. Rolling off their highs (turning down

the treble) makes the rear speakers harder to pinpoint. He also notes they sit far from the windshield; in the same discussion he names windshield reflections as something that narrows the stage.

1.11 #3 Rears must be stereo, never mono or just "front speakers mounted in back".

> Mono rear fill is one of the "bizarro" layouts. Short of channels? Run the front mid and tweeter on one channel with a passive rather than give up stereo rears. RARE
>
> **Sources:** DIYMA MS-8 FAQ p2 · DIYMA 2010-05-27 · DIYMA 2015-08-23

In plain English: Rear fill is sound from rear speakers that supports the front stage. He insists rears be stereo (separate left and right), never mono, and never just "front speakers mounted in back", meaning unprocessed copies of the front sound. He calls mono rear fill one of the "bizarro" layouts. If you are short of amplifier or processor channels, his answer is to run the front midrange and tweeter on one channel with a passive crossover (a small filter that splits one channel between the two speakers) rather than give up stereo rears.

1.11 #4 His rear-fill recipe without an upmixer: plain stereo, high-passed, raised, then delayed until the stage returns to the front.

> Tune the front first. Feed the rears normal stereo with only a high-pass (at least 100 Hz) and EQ out big peaks. Raise them until the stage starts moving rearward, then add delay (~10–20 ms) until it's anchored in front; more level needs more delay, and ~35 ms becomes an echo. RARE
>
> **Sources:** DIYMA 2021-07-25 · DIYMA 2021-12-07 · OFCAP Ep. 19 @3:33:00

In plain English: This is his step-by-step recipe for rear fill without an upmixer. First tune the front speakers. Then feed the rears normal stereo with only a high-pass filter (which blocks the bass) at 100 Hz or higher, and use EQ to remove any big peaks. Raise the rear level until the stage (where the music seems to be) starts moving toward the back. Then add delay, around 10–20 ms (thousandths of a second), until the stage is anchored in front again. The idea rests on a basic fact of hearing: we tend to place a sound where it first arrives from, so slightly late rear sound can add space without pulling the image backward. The more level you give the rears, the more delay they need, and at about 35 ms the rear sound becomes a distinct echo.

1.11 #5 He dropped differential (L−R) rear fill for plain stereo.

> In 2017 he gave a miniDSP recipe for L−R rears delayed 12–20 ms, and says he may have helped give L−R fill its "semi-mythical" status (a member's phrase). By the 2020s he called it nasal, honky and "phasy"; stereo changes the response less and works on nearly all music. Wiring rears out of phase, the pre-DSP trick, cancels bass and center. RARE **Changed view:** 2017 L−R → 2020s in-phase stereo
>
> **Sources:** DIYMA 2017-10-06 · D&F live @33:05 · DIYMA 2021-12-07

In plain English: "Differential" or L−R rear fill feeds the rears the left channel minus the right, which keeps only what differs between them and removes centered sounds. In 2017 he gave a recipe for doing this on a miniDSP (an inexpensive processor) with the rears delayed 12–20 ms. He says he may have helped give L−R fill its "semi-mythical" status (a forum member's phrase). By the 2020s he had dropped it, calling it nasal, honky and "phasy", and he now prefers plain stereo, which

changes the tonal response less and works on nearly all music. The old pre-processor trick of wiring rears out of phase cancels bass and center content. The [Changed view] tag marks this shift from L−R (2017) to normal in-phase stereo (2020s).

1.11 #6 If you still build L−R fill: it's one signal, and leftover center usually means mismatched inputs.

> For L−R on one side and R−L on the other, just reverse one rear speaker's polarity (a bridgeable amp already inverts one channel inside). Center "leaking" into L−R usually comes from test noise that isn't truly mono (or is lossy), unequal L/R EQ or level, or phase differences between head-unit outputs.
>
> **Sources:** DIYMA 2021-11-24 · DIYMA 2022-01-15

In plain English: If you still want L−R rear fill, he gives two practical notes. First, L−R on one side and R−L on the other are the same signal, one simply upside down, so you can just reverse one rear speaker's polarity (swap its wires). A bridgeable amplifier (one that can join two channels into one) already inverts one channel inside. Second, if centered sound "leaks" into the L−R rears, where it should cancel out, the usual causes are mismatched inputs: test noise that isn't truly mono (or is lossy, meaning compressed like an MP3), unequal left/right EQ or level, or phase differences between the head unit's outputs.

1.11 #7 Logic7 rule: high-pass rears and sides at 100 Hz, 24 dB/oct, because one of them gets polarity-flipped.

> To anchor front-steered sounds in front, Logic7 inverts one rear/side channel, which would cancel midbass (which belongs up front anyway). Rears should play 100 Hz to at least 10 kHz with a tweeter ("Coaxials are great for rears"). RARE Era: MS-8/Logic7
>
> **Sources:** DIYMA MS-8 FAQ #11 · DIYMA MS-8 FAQ #16 · DIYMA MS-8 FAQ p2

In plain English: This rule is for Logic7, the Harman upmixer used in the MS-8 era. High-pass the rears and sides at 100 Hz with a steep 24 dB-per-octave slope. The reason: to keep sounds that belong up front anchored in front, Logic7 flips the polarity of one rear or side channel, and if those speakers played midbass, the flipped one would cancel it. The midbass belongs up front anyway. He says rears should play from 100 Hz to at least 10 kHz with a tweeter: "Coaxials are great for rears" (a coaxial has its tweeter mounted in the middle of the woofer).

1.11 #8 Sides at 12 ms, rears at 20 ms, tweeters at ear level.

> Rear-door speakers act as sides and package-tray speakers as rears; mount tweeters high with a clear path to the ears (no aiming needed). He prefers sides over rears (they widen the stage a bit) and ear-level spots like C/D-pillars; 5.1 versus 7.1 is "really hard to hear". RARE
>
> **Sources:** DIYMA 2015-03-03 · DIYMA MS-8 FAQ p3

In plain English: These are his delay and placement figures. In his scheme, the rear-door speakers act as "sides" and the package-tray speakers (on the shelf behind the back seat) act as "rears". Delay the sides by 12 ms and the rears by 20 ms. Mount tweeters high, with a clear path to the ears; no aiming is needed. He prefers sides over rears because they widen the stage a bit, and likes ear-level spots such as the C- or D-pillars (the body posts behind the rear doors). He finds 5.1 versus 7.1 (five versus seven main speaker channels, plus a sub) "really hard to hear".

1.11 #9 Don't buy fancy rear speakers, but do use rear tweeters.

> "Use what you have": matching rear size or timbre to the fronts isn't critical, since rears play mostly processed content at low level. Rear tweeters help even in an "SQ" car; he mocked treating them as a competition secret. RARE
>
> **Sources:** DIYMA 2010-06-01 · DIYMA 2010-05-22 · DIYMA 2021-07-24

In plain English: Many people assume rear speakers must match the fronts in size or sound. He says "Use what you have": because rears play mostly processed content at low level, matching their size or timbre (tonal character) to the fronts isn't critical, so there's no need to buy fancy ones. He does recommend rear tweeters, even in an "SQ" (sound-quality competition) car, and he mocked the idea of treating them as a competition secret.

1.11 #10 Rears are the condiment.

> Front L/R are the bread, the center the meat and cheese, the rears the condiment. Under Logic7 rear level depends entirely on the recording's out-of-phase content, and soloed rears swell and fade "like a wave", so solo them only to diagnose. RARE
>
> **Sources:** DIYMA MS-8 FAQ p3 · DIYMA MS-8 FAQ p3 (solo)

In plain English: His sandwich comparison: the front left and right speakers are the bread, the center is the meat and cheese, and the rears are the condiment. They add flavor but are not the meal. Under Logic7, how loud the rears play depends entirely on how much out-of-phase content the recording has. If you solo them (play them alone), they swell and fade "like a wave". So solo the rears only to diagnose a problem.

1.11 #11 One stage for everyone, not a second stage for the back seat.

> OEM Logic7 gives rear passengers their own stage at the front seatbacks, which he calls "seriously difficult"; he prefers one stage at the dash for every seat. For the back seat, delay the sides to arrive with the fronts there, and raise them only until the front images start moving rearward.
>
> **Sources:** DIYMA MS-8 FAQ #11 · DIYMA 2016-01-16

In plain English: Factory Logic7 systems give back-seat passengers their own stage at the backs of the front seats. He calls that approach "seriously difficult" and prefers one stage, at the dash, for every seat. To make that work for the back seat, delay the side speakers so their sound arrives at the rear seats together with the sound from the fronts. Then raise the sides only until the front images start moving backward.

1.12 Upmixing: matrix decoders, factory upmixers and music upmixers

What this part is about: An upmixer is a processor that takes ordinary two-channel stereo and spreads it across more speakers: a center, sides and rears. This part explains the different kinds (older "matrix" decoders made for movies, the ones built into factory systems, and newer music-focused ones), how they work and fail, how to check and set them, and his long experience living with them.

1.12 #1 How matrix upmixers (Pro Logic II, Logic7) steer.

> An upmixer spreads stereo over more channels. Matrix types compute a steering angle thousands of times a second and ride whole-channel gains: left/center/right from level differences, front/rear from phase (in-phase forward, out-of-phase rearward). They were built to decode surround hidden in 2-channel film media like VHS. RARE
>
> **Sources:** DIYMA MS-8 FAQ #11 · DIYMA 2018-04-06 · DIYMA 2017-07-01

In plain English: An upmixer spreads stereo over more channels. "Matrix" types, such as Dolby Pro Logic II and Harman's Logic7, work out a "steering angle" (an estimate of where the main sound is coming from) thousands of times a second. They then ride, meaning turn up and down, the volume of whole channels. Left/center/right is decided from level differences: a sound louder in the left channel goes left, and one equal in both goes to the center. Front/rear is decided from phase: in-phase sound (left and right moving together) goes forward, out-of-phase sound (moving opposite) goes rearward. He notes they were built to decode surround sound hidden inside two-channel film media such as VHS tapes.

1.12 #2 Check the steering with pink noise.

> Mono pink noise should come mostly from the center (L/R about 6 dB down); invert one channel and it moves to the rears; balance hard right should play only the right front. Stereo pink noise, with the center level set by the mono share, is probably the best check. RARE
>
> **Sources:** DIYMA 2017-08-05 · DIYMA MS-8 FAQ #25

In plain English: Pink noise is a steady hiss containing all frequencies, used for testing. He gives simple checks that a matrix upmixer is steering properly. Mono pink noise (identical in both channels) should come mostly from the center, with left and right about 6 dB down (clearly quieter). Invert one channel (flip its polarity) and the noise should move to the rears. Turn the balance hard right and only the right front should play. Stereo pink noise (different noise in each channel) plays through all the speakers, with the center's level depending on how much of the content is mono; he calls this probably the best check.

1.12 #3 Where matrix upmixing fails, and what breaks it.

> It can't steer a transient one way while a steady sound goes another (speech in front kills steady rain in the rears). Weak FM, factory per-band delays and FIR processing look like phase shifts and steer strangely, as do producers' out-of-phase tricks and lossy files. Remove factory delay, not just EQ, and never stack two upmixers. RARE
>
> **Sources:** DIYMA MS-8 FAQ #16 · DIYMA 2018-04-07 · EDUCAR

In plain English: Because a matrix upmixer turns whole channels up and down, it can't send a short, sharp sound one way while a steady sound goes another. His example: speech in front kills steady rain in the rears, because the speech pulls the steering forward. Anything that looks like a phase shift fools it and makes it steer strangely: weak FM radio reception, factory delays applied per frequency band, and FIR processing. So do producers' out-of-phase studio tricks and lossy files (compressed formats like MP3). His practical rules: when feeding it a factory signal, remove the factory delay, not just the factory EQ, and never stack two upmixers one after the other.

1.12 #4 Feed an upmixer two channels, not discrete 5.1, and use 5.1 outputs rather than 3.1.

> Logic7 builds center, sides and rears from stereo; for DVDs pick the 2-channel downmix. Home 5.1 assumes a TV, dialogue and near-equilateral seating, none of which fits a car; upmixing is not just for movies. RARE
>
> **Sources:** DIYMA MS-8 FAQ #8 · DIYMA 2018-06-04

In plain English: "Discrete 5.1" is a movie soundtrack already split into separate channels: front left, center, front right, two surrounds and a sub. He says to feed an upmixer two-channel stereo instead, and to use its 5.1 outputs (including sides or rears) rather than only 3.1 (left, center, right and sub). Logic7 builds the center, sides and rears from stereo, so for DVDs pick the two-channel downmix. Home 5.1 assumes a TV screen, dialogue, and seating that forms a near-equilateral triangle with the speakers, and none of that fits a car. His point is that upmixing is not just for movies.

1.12 #5 When to switch Logic7 on.

> No center and no rears: off ("it does nothing for you"). Rears but no center: try it. Center but no rears: add rears. In MS-8 the center plays only with Logic7 on, so if you dislike rear steering, leave it on and fade forward. Era: MS-8
>
> **Sources:** DIYMA MS-8 FAQ p2 · DIYMA MS-8 FAQ p3

In plain English: This is MS-8-era advice on when to switch on its Logic7 upmixer. No center and no rears: leave it off ("it does nothing for you"). Rears but no center: try it. A center but no rears: add rears. In the MS-8 the center speaker plays only when Logic7 is on, so if you dislike sound being steered to the rears, leave Logic7 on and use the fader to move the sound forward.

1.12 #6 Matrix upmixers for music: from "the ticket" to poor.

> MS-8 era: for 2-channel music a matrix processor "is the ticket", more believable than Audyssey's "more clinical" sound. Later: PL, PLII and Logic7 suit video but steer mixers' tricks rearward in music; music-oriented upmixers (Circle Surround, Harman QLS, Audiofrog Multiseat) extract common content instead. RARE Changed view: 2008–10 → 2017 onward
>
> **Sources:** DIYMA MS-8 FAQ #10 · DIYMA 2017-10-12 · D&F live @35:06

In plain English: His view of matrix upmixers for music changed over time, which the [Changed view] tag marks. In the MS-8 era (2008–10) he said that for two-channel music a matrix processor "is the ticket", more believable than the "more clinical" sound of Audyssey (the technology in Alpine's Imprint processors then). From 2017 on, he said PL, PLII and Logic7 suit video but, with music, steer mixing engineers' out-of-phase tricks to the rear. He contrasted them with music-oriented upmixers (Circle Surround, Harman QLS, Audiofrog Multiseat), which instead pull out the content common to both channels.

1.12 #7 What a music upmixer should do.

> Use clues in the recording to make it more believable without altering it, only spreading it across channels; the one standard he knows (an ITU broadcast rule) requires the upmix to fold back to stereo, which matrix processors can't. A good one matches or beats stereo in more than one seat but rarely equals stereo in its one ideal seat. RARE
>
> **Sources:** D&F live @41:05 · DIYMA 2017-10-12 · EDUCAR

In plain English: He sets out the job description. A music upmixer should use clues already in the recording to make it more believable, without altering the recording, only spreading it across more channels. The one standard he knows of, a broadcasting rule from the ITU (International Telecommunication Union), requires that the upmix fold back to the original stereo if you mix all its channels together again. Matrix processors can't do that. In his view a good upmixer matches or beats stereo in more than one seat, but rarely equals stereo in stereo's one ideal seat.

1.12 #8 Audiofrog Multiseat Stereo splits rather than steers.

> Built on Penteo (a studio upmixer) and reworked for cars, it splits stereo into mono (to the center), similar-but-not-identical L/R, and L-only/R-only content, spreading the last two over L/R, sides and rears "like a horseshoe". With no phase-based steering there's no coloration, bass stays punchy, and every seat works without L/R delays. RARE
>
> **Sources:** DIYMA 2017-11-13 · DIYMA 2018-04-06 · Mobile Electronics Aug 2016

In plain English: Audiofrog is the company Andy co-founded, and Multiseat Stereo is its upmixer. It is built on Penteo (an upmixer made for recording studios) and reworked for cars. Instead of steering whole channels, it splits stereo into three parts: mono content (identical in both channels), sent to the center; content that is similar but not identical in left and right; and content found only in the left or only in the right. The last two are spread over the left and right fronts, sides and rears "like a horseshoe". He says that because no phase-based steering is used, there's no coloration (no change to the tonal balance), bass stays punchy, and every seat works without left/right delays.

1.12 #9 He has lived with upmixers for 20+ years.

> His cars have run Logic7, PLII, DTS Neural, a custom room-synthesis algorithm and Audiofrog's upmixer; Audiofrog's CES 2016 car (DTS Neural plus a real center) so that "the driver's seat sounds just like the passenger seat". At home his JBL M2 monitors play multichannel 95% of the time. RARE
>
> **Sources:** CES 2016 @3:03 · DIYMA 2015-08-09 · DIYMA 2021-09-08

In plain English: This point is about his own experience. Over more than 20 years his cars have run Logic7, PLII, DTS Neural, a custom room-synthesis algorithm (processing that creates the sense of a room) and Audiofrog's own upmixer. Audiofrog's demo car at CES 2016 (a big electronics trade show) used DTS Neural plus a real center so that "the driver's seat sounds just like the passenger seat". At home, his JBL M2 studio monitors play multichannel 95% of the time.

1.12 #10 Factory upmixers (e.g. Harman QLS): use them instead of ripping them out.

> QLS (Hyundai/Kia, Maserati, Lincoln) separates center, left, right and in-between content with about seven filters, and its outputs may not sum back to stereo. Wire each factory output to a matching amp channel, add power and better speakers, then just re-EQ and set levels; the delays and crossovers are already there. RARE
>
> **Sources:** EDUCAR · Absolute Electronix 2023 @58:34 · Absolute Electronix 2023 @56:27

In plain English: Many installers believe the first step in upgrading a premium factory system is to rip out or bypass its processing. For factory upmixers, he says the opposite. Harman's QLS (found in Hyundai/Kia, Maserati and Lincoln systems) separates center, left, right and in-between

content with about seven filters, and its outputs may not add back up to plain stereo. So connect each factory output to its own matching amplifier channel, add power and better speakers, then simply re-EQ and set levels. The delays and crossovers are already in place.

1.12 #11 Upmixed cars are harder to tune: L+R must sum like center plus L−R.

> Harman's factory tuners use mic arrays and in-house software to meet that. RARE
>
> **Sources:** DIYMA 2012-11-02

In plain English: He says a car with an upmixer is harder to tune than a plain stereo car. The combined sound of the left and right speakers (L+R) has to match closely the combined sound of the center (the mono part) plus the L−R (difference) part. Harman's factory tuners use arrays of several microphones and in-house software to meet that. The source states the requirement without spelling out the reasoning further.

1.12 #12 His two-seat (and back-seat) process with an upmixer (2016).

> Front delays to a point midway between the front headrests; crossovers by dispersion and safety; measure each side plus sub in both front seats (six mic spots on a 7" circle), average and auto-EQ; repeat for center and surrounds; delay sides to arrive with the fronts at the rear seats; add 12–20 ms to the rears; set the upmixer for the widest stage with a stable center. RARE
>
> **Sources:** DIYMA 2016-01-16

In plain English: This is his 2016 process for tuning two front seats, plus the back seat, with an upmixer. Step by step: (1) Set the front delays to a point midway between the two front headrests, so neither seat is favored. (2) Choose crossovers by dispersion (how widely each speaker spreads its sound) and by safety (keeping each driver away from notes that could damage it). (3) Measure each side plus the sub from both front seats, at six mic spots on a 7-inch (about 18 cm) circle, average the results and let the auto-EQ correct them. (4) Repeat for the center and the surround speakers. (5) Delay the side speakers so their sound reaches the rear seats together with the fronts. (6) Add 12–20 ms (thousandths of a second) of delay to the rears. (7) Set the upmixer for the widest stage that still keeps a stable center.

1.12 #13 Room synthesis fooled the judges.

> At his last IASCA/MECA show judges said they'd never heard a car reproduce a recording's room so accurately; his DSP generated it. He tolerates "HUGE errors in frequency response" when spatial qualities are good. RARE
>
> **Sources:** DIYMA MS-8 FAQ p4 #70 · DIYMA MS-8 FAQ p5 #99

In plain English: IASCA and MECA are car-audio competitions. At his last such show, he says, judges told him they had never heard a car reproduce a recording's room (the sense of the space it was recorded in) so accurately. In fact his processor was generating that room, not just playing it back. He says he tolerates "HUGE errors in frequency response" (unevenness in tonal balance) as long as the spatial qualities (width, depth, sense of space) are good.

1.12 #14 Aftermarket DSPs lack upmixers for business reasons, not demand.

> Licenses are cheap, but a product manager must understand upmixing and win it in a meeting that includes finance; many are steeped in 2-channel audio or judged it from bad setups. Nearly every new premium factory system has one. RARE
>
> **Sources:** DIYMA 2017-07-07 · DIYMA 2021-08-01

In plain English: Aftermarket DSPs are the add-on sound processors sold to car-audio enthusiasts. Most don't include an upmixer, and he says that is for business reasons, not lack of demand. Licenses for the technology are cheap, but a product manager must understand upmixing and win the argument for it in a meeting that includes finance. Many product managers are steeped in two-channel audio or judged upmixing from badly set-up systems. Meanwhile, he notes, nearly every new premium factory system has an upmixer.

1.13 Believability over accuracy

What this part is about: This part sums up his philosophy. Because a recording is not a true copy of a real event and a car is not a concert hall, he aims for sound you can believe in rather than "accuracy", but built on getting the basics right first.

1.13 #1 "Think of it as believability rather than accuracy."

> The guitar never travels through the circuit board, only a signal, which must be changed to make up for the acoustic system; the reward for choosing believability "might be greater than you expect". RARE
>
> **Sources:** DIYMA MS-8 FAQ #13 · DIYMA Apr 2010

In plain English: Many people believe the goal of a good system is "accuracy": passing the recorded signal through untouched. He suggests thinking of it as believability instead. His reasoning: the guitar itself never travels through the circuit board, only an electrical signal that stands for it. That signal has to be changed (equalized, delayed and so on) to make up for the acoustic system, meaning the speakers and the car interior they play into. He says the reward for choosing believability "might be greater than you expect".

1.13 #2 There's no accurate reference to be accurate to.

> Most records are many events sped up, pitch-shifted, compressed and given a synthesized room; a studio disc is a "deep fake" of an event, so aim for suspension of disbelief. Producers want the studio experience, not "flat": with his DSP off they asked why their tracks sounded "mono"; with it on, "it needs more bass". RARE
>
> **Sources:** DIYMA MS-8 FAQ p5 #83 · DIYMA MS-8 FAQ p4 #70 · CMA Networks 2023 @35:05

In plain English: He argues that most records have no real original to be accurate to. They are built from many separate recordings that have been sped up, pitch-shifted, compressed (their loud and quiet parts squeezed closer together) and given a synthesized, artificial room. He calls a studio disc a "deep fake" of an event, so the aim is suspension of disbelief: letting you accept the illusion. He adds that producers want to hear the studio experience, not "flat" (equal loudness at every frequency). When he played producers their tracks with his processor off, they asked why the tracks sounded "mono"; with it on, they said "it needs more bass".

1.13 #3 Two-channel sounds like a diorama on the dash; multichannel sounds like a space bigger than the car, for two people at once.

> He points to Apple spatial audio and Dolby Atmos, and to a 120-plus-speaker L-ISA concert he heard as "a comforter of sound": a non-stereo presentation can be beautiful. Aftermarket listeners judging factory premium systems only on bass and loudness miss exactly this. RARE
>
> **Sources:** Absolute Electronix 2023 @55:28 · CMA Connected Oct 2021 @48:18 · CMA Networks 2023 @39:03

In plain English: A diorama is a small model scene. He says plain two-channel stereo in a car sounds like a diorama sitting on the dash, while multichannel sound can feel like a space bigger than the car, and for two people at once. He points to Apple spatial audio and Dolby Atmos (modern surround formats), and to a concert using L-ISA (a multi-speaker concert sound system) with more than 120 speakers, which he heard as "a comforter of sound", like a thick quilt wrapped around you. His point is that a non-stereo presentation can be beautiful. Aftermarket listeners who judge premium factory systems only on bass and loudness miss exactly this.

1.13 #4 Believability rests on the basics, not instead of them.

> He values width most, then ambience; a peaky midrange is forgivable, a thick midbass is his pet peeve. Experienced listeners judge a demo car in seconds: full, impactful bass from the front, a centered singer, then width and depth. For customers, first "get close to correct".
>
> **Sources:** OFCAP Ep. 19 @3:31:01 · Absolute Electronix 2023 @53:21 · OFCAP Ep. 38 @54:01

In plain English: He stresses that believability is built on the basics, not a replacement for them. Width matters most to him, then ambience (the sense of the recording's space). He can forgive a peaky midrange (some middle notes sticking out), but a thick midbass (too much low-middle sound) is his pet peeve. He says experienced listeners judge a demo car within seconds: first full, impactful bass that seems to come from the front, then a centered singer, then width and depth. So in practice, for customers, the first job is to "get close to correct".

1.14 Top rarely-discussed points in this section

- Door speakers sit 180° out of phase near 250 Hz in both front seats, right in the vocal band (1.1 #3, 1.9 #4).
- Delay never moves a speaker; center an off-center image with level, never with more delay (1.1 #7, 1.1 #8).
- Set delays to the midrange's distance, and let crossover phase, not pillar mounting, set vocal height (1.1 #6, 1.1 #12, 1.7 #7).
- Above the midbass, left plus right sums to only about +3 dB in a car; leave anything within ±90° alone (1.1 #13).
- A car has no reverberant field: you hear the power response, and no EQ or delay fixes a reflection comb (1.3 #1, 1.3 #4).
- Predict reflection nulls with a tape measure; push mids into corners and move tweeters away from surfaces (1.3 #5, 1.6 #8).
- Dash mats and foam do nothing in the midrange; ~200 Hz would need ~1.5 ft of absorber (1.3 #8).

- Dispersion is diameter versus wavelength, and his circumference ÷ wavelength rule decides where each driver can go (1.4 #1, 1.4 #2).
- A 6" + 1" 2-way has an off-axis hole that aiming only moves, which is why cars need 3-ways when studios don't (1.4 #4, 1.4 #6).
- Aim only the tweeter; door midbass aiming changes "zip" (1.5 #1, 1.5 #2).
- A small mid high in the door beats A-pillar pods, which comb the whole midrange and are one-seat-only without a center (1.6 #2, 1.6 #6, 1.9 #3).
- "HRTF height EQ" works only while you're adjusting it (1.2 #5).
- Images can't be made reliably outside the speaker triangle; "width" from delay games is comb filtering (1.7 #1, 1.7 #2).
- Two-seat stereo without a center: an all-pass near 250 Hz instead of left/right delay, as Toyota, Subaru and Bose already do (1.9 #4, 1.9 #5).
- Two listening tests reveal whether a factory system uses all-pass filters or delay (1.9 #6).
- L+R is not a center; real extraction leaves center content in L/R about 6 dB down (1.10 #1, 1.10 #2).
- A dash center's ~2 kHz windshield dip is a feature that mimics a phantom center (1.10 #5).
- Keep a factory upmixer and go one-to-one instead of "getting back to flat" (1.12 #10).

Changed views and conflicts in this section

- **Windshield corner:** 2012 "keep speakers away from the junction" → 2013/2021 "tight in the corner is best; a pod hanging nearby is the problem" (1.6 #8).
- **Firing mids into the glass:** did it himself in 2009 → from 2011 it narrows the stage, and in Sep 2021 up-firing mids make the densest comb; conflict with his Oct 2021–Jan 2022 praise of windshield-aimed factory dash "middlers" (1.6 #9, 1.6 #23).
- **Tweeterless widebanders:** 2012–14 "must be on-axis" → 2018–21 "aim at the windshield; on-axis sounds clinical" (1.8 #4).
- **Phase EQ for two seats:** 2013–15 inaudible next to delay → Jan 2016 "drop L/R delay for phase EQ" → 2021 helps "only with fiddling", so he won't teach it (1.9 #4, 1.9 #10).
- **Rear fill:** 2017 differential (L–R) recipe → 2020s plain in-phase stereo, high-passed and delayed (1.11 #5).
- **Matrix upmixers for music:** 2008–10 "the ticket" → from 2017 poor for music; music-oriented upmixers instead (1.12 #6).
- **Stage height:** 2016 "from physical location" → 2020–21 adds that crossover phase holds everything at tweeter height (1.7 #7).
- **Conflict, A-pillars:** "a HORRIBLE place for a midrange" versus Audiofrog copy calling GB10+GB25+GB60 "the ultimate active system for A-pillars" and a GB25 designed for pillar pods (1.6 #6, 1.6 #7).
- **Conflict, crossfired pillar tweeters:** his "band-aid" verdict versus a 2015 Harman/JBL/Infinity tip sheet that recommends it and the Bose Fiat 500 he praised (1.5 #6).

- **Conflict, Harman patent:** he recalls US6876748 naming 250 Hz; its text cites about 200 Hz (1.9 #5).
- **Unchanged:** "HRTF EQ" can't raise the stage, from 2008 through 2021 (1.2 #5).

SECTION 2

2. Installation craft and hardware choices

Andy Wehmeyer's installation advice is applied physics: put speakers where the car helps rather than hurts, mount them solid and sealed, cross every driver where it is safe and still disperses widely, and stop spending effort (door boxes, foam, aiming, show fabrication) that doesn't change what reaches your ears. Points come from his DIYMobileAudio (DIYMA) posts (anonymized accounts "14642", 2008–2016, and "156546", 2017–2022, both his), Audiofrog's blog, manuals and product pages ("COMPANY" marks unbylined company text), and his podcasts and videos. Jargon used throughout: **IB (infinite baffle)** = a speaker mounted in a large closed space such as a door, with no box; **Fs** = a driver's free-air resonance (**Fc** once it is in a box); **Q/Qts** = how peaky a driver is around resonance; **Vas** = the air volume equal to its suspension stiffness; **Xmax** = linear excursion; **LR4** = a 24 dB/octave Linkwitz-Riley crossover.

What this section is about: This section covers the hands-on side of a car system: where the speakers go, how they are mounted, what sound-deadening material really does, and which hardware suits which spot. His approach is practical physics: put speakers where the car helps rather than hurts, mount them solid and sealed, cross each one over where it is safe, and skip work that doesn't change what reaches your ears. Where a point is marked COMPANY, the words come from unsigned text by Audiofrog (his company), not from him by name. A few terms used throughout: Hz (hertz) measures pitch, with low numbers meaning deep bass; dB (decibels) measures loudness, and +6 dB means double the sound pressure, an easily heard increase; a high-pass filter is a crossover setting that stops low notes from reaching a speaker.

2.1 Doors and midbass: baffle, not box

What this part is about: How to mount and choose the mid-size bass speakers ("midbass", usually 6 to 8 inches) that sit in car doors. His message is that a door needs a good seal rather than a box, that the door and the car's corners already add bass for free, and that the deepest bass belongs to a subwoofer.

2.1 #1 A door midbass needs a sealed baffle, not a box.

> RARE With any high-pass, a door box is "a waste of time and money": the electronic filter limits excursion as a box's air spring would, so you only need front and back "pretty well isolated". Seal the speaker's front to the trim panel, or put a couple of layers of Dynamat over the sheet-metal holes (both is fine). A door box's only benefit is stopping rattles; if you build one, size it like a sub box.
>
> **Sources:** DIYMA 2009-03-29 · DIYMA 2008-11-22 · DIYMA 2015-08-19

In plain English: A baffle is the surface a speaker is mounted on. Its job is to keep the sound from the back of the cone apart from the sound from the front, because the two are opposite and cancel when they mix. A box does that too, and its trapped air also acts like a spring that limits how far the cone moves. His point is that once you use any high-pass filter, the filter already limits the cone's movement, so the door needs no box. Front and back just need to be "pretty well isolated". You can seal the speaker's front to the door's trim panel, or cover the holes in the door's sheet

metal with a couple of layers of Dynamat (a sound-deadening sheet). Doing both is fine. Many people believe door midbass needs a box. He calls one "a waste of time and money" whose only benefit is stopping rattles, and says that if you build one, size it as carefully as a subwoofer box.

2.1 #2 Tiny sealed boxes behind midbass drivers are "bass prevention".

> RARE A small box raises the resonance and Q, cuts low end, needs more power for the same bass and adds a peak at resonance. A high-pass limits excursion just as well, and EQ supplies any boost the box would have given.
>
> **Sources:** DIYMA 2014-01-30 · DIYMA 2010-07-04

In plain English: Every speaker has a resonance, a frequency where its cone moves most easily. Below that, its bass output falls away. "Q" describes how peaky the sound is around that resonance. A small sealed box adds the stiffness of its trapped air, which pushes the resonance and the Q up. He says this cuts the deep end, makes the speaker need more amplifier power for the same bass, and adds a bump (a peak) at the new resonance. That is why he calls tiny boxes "bass prevention". Many people box midbass drivers to protect them. His point is that a high-pass filter limits the cone's movement just as well, and EQ (equalization, the electronic tone controls in a processor) can add back any boost the box would have given.

2.1 #3 In a door, air volume and sealing matter; Fs and Q barely do once the driver is high-passed.

> RARE A door's volume is similar to or larger than a car driver's Vas, so it hardly changes the driver (IB-type drivers have Qts about 0.5–0.8). With a high-pass above Fs plus EQ, "does it suit IB?" debates are pointless. A mid and a midbass can even share one door cavity: it is huge and "really lossy".
>
> **Sources:** DIYMA 2015-10-10 · DIYMA 2015-07-24 · DIYMA 2014-04-01

In plain English: Vas is the amount of air that is exactly as springy as the speaker's own suspension. A space much smaller than Vas stiffens the speaker and changes how it behaves; a space as big or bigger barely does. He points out that a car door holds about as much air as a typical car speaker's Vas, or more, so the door hardly changes the driver. Drivers meant for infinite-baffle (no-box) use have a Qts of about 0.5–0.8. So once the speaker has a high-pass set above its resonance (Fs) and some EQ, arguing over whether a driver "suits IB" is pointless. What matters is the door's air volume and how well it is sealed. A midrange and a midbass can even share one door cavity. He says no good engineer would do that in a home speaker, but in a car it is no big deal: a door is huge and "really lossy" (its air does not act as a tight spring).

2.1 #4 The door doesn't change the speaker; the cabin changes what reaches you.

> A driver flat from 100 Hz to 10 kHz stays flat in a door, and no driver can be designed to cancel the car: the gap from its anechoic curve is reflections, which EQ handles. So pick drivers for the job their data shows (he says a Purifi 4" is plainly built for a small vented home 2-way).
>
> **Sources:** DIYMA 2013-10-30 · DIYMA 2021-12-24

In plain English: An "anechoic" measurement is taken in a room with no echoes, so it shows the speaker's own sound. His point is that mounting a speaker in a door does not change the speaker. If it measures flat (even) from 100 Hz to 10 kHz, it is still flat in the door. When your in-car measurement looks different, the difference is the car's reflections, and EQ is the tool for those.

No speaker can be designed to cancel out what a car does. So pick a driver for the job its published data shows it was designed for. His example is the Purifi 4", which he says is plainly built for a small vented two-way home speaker.

2.1 #5 Ported midbass: generally no, and never vent a door box into the door.

> A port big enough to avoid noise is "huge", and a grille in front of it shifts the tuning; a long-coil sealed/IB driver with power and EQ is easier. The one exception: a high-Fs driver you are forced to use, vented to lower its −3 dB point. A port firing into the door throws its output away, leaving less bass than sealed.
>
> **Sources:** DIYMA 2011-10-18 · DIYMA 2008-11-22 · DIYMA 2022-01-13

In plain English: A ported (vented) box has an opening, the port, that adds bass around a chosen "tuning" frequency; a sealed box has no opening. He is generally against porting midbass. A port big enough not to make air noise would be "huge" for a midbass, and a grille in front of the port shifts its tuning. He finds it easier to use a sealed or IB driver with a long voice coil (so it can move further) plus enough power and EQ. His one exception is a driver with a high resonance (high Fs) that you are forced to use, for example because nothing else fits. A vent can lower its −3 dB point, the frequency where its bass has dropped to half power. Never vent a door box into the door. What you hear from a ported box is the speaker plus the port, so a port firing into the door throws its output away, leaving less bass than a sealed box would give.

2.1 #6 Pull out foam cup "speaker baffles"; at most keep the top as a rain shield.

> RARE For a "lifeless and hollow" system his first instruction was to remove the foam baffles (then check polarity, set gains and recalibrate). In 2021: cut off the bottom of the cup and keep only the upper rain-shield part.
>
> **Sources:** DIYMA 2013-05-19 · DIYMA 2021-09-21

In plain English: Foam "speaker baffles" are foam cups sold to sit behind a car speaker. A forum member described his system as "lifeless and hollow". The first thing Andy told him to do was take those foam baffles out. After that: check polarity (that every speaker's + and − are wired the same way round), set the amplifier gains (input levels), and recalibrate (rerun the processor's setup). In 2021 his advice was to cut off the bottom of the cup and keep only its top part, as a shield against rain water running down inside the door. The sources don't spell out his reason.

2.1 #7 Foam rings go on the bracket, never on the speaker; they stop rattles, they don't "make midbass".

> RARE Returned speakers often arrive caked in foam-ring adhesive, sometimes stuck to the surround. His adapters carry closed-cell foam about 3/4" wide on the bracket, never touching the speaker, to stop the door panel rattling.
>
> **Sources:** OFCAP Ep. 38 @34:02

In plain English: Foam rings are strips of foam placed around a car speaker. Many people believe they "make midbass". He says their only job is to stop rattles. Never stick them to the speaker itself. Speakers sent back to his company often arrive caked in the rings' glue, sometimes even stuck to

the surround (the flexible edge around the cone). Audiofrog's mounting adapters instead carry closed-cell foam (a dense foam whose bubbles are sealed) about 3/4 inch (19 mm) wide on the bracket. It never touches the speaker, and it stops the door panel rattling against the bracket.

2.1 #8 The door and the car's corner give midbass its bass: about +6 dB per halving of the radiating space.

> RARE Below about 500 Hz a 6" radiates everywhere; the baffle reflects its rearward-going front sound forward (+6 dB) and a corner adds another +6 dB. Midbass always sits near a corner (door/floor or kick panel), so the location boosts bass on top of cabin gain.
>
> **Sources:** Audiofrog blog Jun 2021 · DIYMA 2017-08-17

In plain English: Low notes spread out from a speaker in every direction, like ripples spreading in a bathtub. If a wall blocks half of that space, the sound that would have gone backward is bounced forward, so it gets louder in front. His rule of thumb is about +6 dB each time the space the speaker radiates into is cut in half. Below about 500 Hz, a 6" speaker radiates everywhere. The door (its baffle) reflects the part heading backward toward the front (+6 dB), and a corner adds another +6 dB. A midbass always sits near a corner (where the door meets the floor, or in a kick panel), so its location boosts its bass. That comes on top of "cabin gain", the extra deep bass a small closed space like a car naturally adds.

2.1 #9 So choose a flat, low-Q midbass; that is why the GB60 has no bass hump.

> RARE A high-Q "European sound" hump helps in a home box but becomes a peak in a car corner, and high Q raises distortion at Fs; flat midbass also blends more easily with the sub (a reviewer measured Qts 0.452, Fs 54.8 Hz). Don't chase ultra-low-Fs "door subwoofer" midbass if you have a sub: one was nearly 10 dB less sensitive in the midbass, "a lot of power".
>
> **Sources:** Audiofrog blog Jun 2021 · DIYMA 2017-08-17 · DIYMA 2021-11-02

In plain English: Some speakers have a high Q, which gives them a bump of extra bass near their resonance. He calls it the "European sound", and it helps in a home speaker box. His point is that in a car corner, which already adds bass (see the point above), that bump turns into an unwanted peak, and a high Q also raises distortion near the resonance (Fs). A flat, low-Q midbass also blends more easily with the subwoofer. That is why Audiofrog's GB60 is designed with no bass hump; a reviewer measured its Qts at 0.452 and Fs at 54.8 Hz. If you have a sub, don't chase "door subwoofer" midbass drivers with an ultra-low Fs. He judged one nearly 10 dB less sensitive in the midbass, which means about ten times the amplifier power for the same loudness: "a lot of power".

2.1 #10 Cross door midbass at about 80 Hz and the sub at about 60 Hz, both LR4; 40 Hz is the worst mistake he sees.

> RARE Door drivers' Fs is around 60–70 Hz; below resonance only the spider controls the cone, distortion soars, the spider wears "like an old shoe" and the car becomes "virtually untunable". The half-octave gap keeps sub and midbass in phase after the 9–12 dB sub boost cars need; at 80 Hz a GB60 moves under 3 mm at 115 W.
>
> **Sources:** OFCAP Ep. 19 @1:45:06 · DIYMA 2020-12-30 · DIYMA 2021-07-24

In plain English: A crossover frequency is where one speaker hands the music over to another. LR4 is a common, steep type of crossover filter. The spider is the fabric ring behind the cone that holds it centered and acts as part of its spring. He says door speakers resonate around 60–70 Hz. Below that resonance only the spider controls the cone, so distortion soars, the spider wears out "like an old shoe", and the car becomes "virtually untunable". That is why crossing door midbass at 40 Hz is the worst mistake he sees. Instead, cross the midbass at about 80 Hz and the sub at about 60 Hz, both LR4. Cars need the sub boosted 9–12 dB, and the half-octave gap (an octave is a doubling of frequency) keeps sub and midbass in phase (pushing air together, so they add rather than cancel) after that boost. At 80 Hz a GB60 moves less than 3 mm at 115 W.

2.1 #11 Door woofers can't replace a sub in a loud system.

> RARE Cars need about 6–12 dB more bass than midrange; getting it from doors means big boosts near 50 Hz, "exactly what your door drivers hate", and the coil leaves the gap (odd-order distortion). Add a 10" sub: if space is short, swap the spare tire for Fix-a-Flat and build in the well, or vent a bandpass box into the trunk. One driver from sub-bass to 1 kHz can't be built and isn't needed.
>
> **Sources:** DIYMA 2013-05-22 · OFCAP Ep. 19 @1:48:08 · DIYMA 2017-07-05

In plain English: He says cars need about 6–12 dB more bass than midrange. Getting that from the door speakers means big boosts near 50 Hz, "exactly what your door drivers hate". The voice coil is pushed out of the magnet's gap, which causes "odd-order" distortion (a kind that adds tones at odd multiples of the note, such as three times it). His fix is to add a 10" subwoofer. If you are short of space, he suggests replacing the spare tire with a can of Fix-a-Flat (tire sealant) and building the sub box in the spare-tire well. Another option is a bandpass box (one where the sound leaves only through a port) that vents into the trunk. He adds that one driver playing from deep bass up to 1 kHz can't be built, and isn't needed.

2.1 #12 No need for 8" or 10" door woofers; with a sub, put a 6" high in the door's top corner.

> RARE With subs low-passed at 100 Hz a user heard no difference at full volume with door panels removed ("Nope, no need"); in 25 years Andy heard one good car with very large front woofers. A 6" in the top corner (the factory grille area) makes left and right far more alike; with correct crossovers most of the image then sits at tweeter height.
>
> **Sources:** DIYMA 2011-08-28 · DIYMA 2021-12-03

In plain English: Many people think bigger door speakers (8" or 10") give better front bass. In the forum thread, a user had his subs low-passed at 100 Hz, meaning the subs played everything below 100 Hz. He took his door panels off and heard no difference at full volume. Andy agreed: "Nope, no need" for 10" door woofers. Andy said that in 25 years he had heard only one good-sounding car with very large front woofers. With a sub, he suggests putting a 6" high, in the door's top corner, around where the factory grille usually sits. That makes the sound from the left and right speakers far more alike. With correct crossovers, most of the stereo image (where you hear the music coming from) then sits at the height of the tweeters.

2.1 #13 Bigger isn't "slower", and punch doesn't live in the midbass.

> RARE The motor, not cone size, sets "speed". Punchy midbass depends on flat response from about 1 to 4 kHz (impact near 3 kHz: a bass player's thumb slap, the snap of toms). Test it on headphones through a 100 Hz, then a 200 Hz low-pass: there is no punch down there.
>
> **Sources:** DIYMA 2011-08-27 · DIYMA 2015-05-09 · DIYMA 2021-10-13

In plain English: Many people believe big woofers are "slow" and small ones are "fast" or punchy. He says cone size doesn't set "speed"; the motor (the magnet and voice coil that drive the cone) does. And the "punch" people want from midbass actually depends on even response from about 1 to 4 kHz. The sense of impact sits near 3 kHz: a bass player's thumb slap, the snap of tom drums. His test is to listen on headphones through a 100 Hz low-pass filter, then a 200 Hz one, so you hear only the notes below that point. There is no punch down there.

2.1 #14 A midbass and a sub are optimized differently; a "sub that plays midrange" mostly doesn't exist.

> RARE A good midbass has low moving mass, about 1/4 a sub's inductance and a very compliant suspension for low Fs without added mass, so it suits a door or rear deck, not a small box. Low Fs plus high efficiency means a huge Vas and a fragile driver (Hoffman's Iron Law); the closest thing, for horn systems, is a large pro woofer in a vented box.
>
> **Sources:** DIYMA 2022-01-13 · DIYMA 2020-08-24

In plain English: Moving mass is the weight of the cone and coil. Inductance is an electrical property of the coil that holds back higher notes. Compliance is how soft the suspension is. He says a good midbass has low moving mass, about a quarter of a sub's inductance, and a very soft suspension. That gives it a low resonance (Fs) without adding weight, but it suits a door or rear deck (big air spaces), not a small box. A driver that has both a low Fs and high efficiency needs a huge Vas and ends up fragile. He ties this to Hoffman's Iron Law, a speaker-design rule that you cannot have deep bass, a small box and high efficiency all at once. The nearest thing to a "sub that plays midrange", used in horn systems, is a large professional (pro audio) woofer in a vented box.

2.1 #15 An 8" in a door is a woofer, not a midrange: the GS8ND2 case.

> RARE About 3" deep for 6×9/8" door openings and tiny boxes (about 7 L sealed, 14 L vented, or IB: under seats, Lexus decks); a shorting ring lets it play to about 300 Hz, so cross it there and add a GS25/GB25. Dual 2 Ω coils: one woofer wired to 4 Ω suits a bridged 150–200 W channel pair; a pair at 2 Ω suits a 500–750 W sub amp. **Changed view:** in 2016 Audiofrog would make no 8" and offered the GS690 6×9 instead; the GS8ND2 arrived in 2020, framed as a woofer.
>
> **Sources:** DIYMA 2016-06-21 · DIYMA 2020-06-24 · CMA Networks 2021 @39:01

In plain English: The GS8ND2 is an Audiofrog 8" speaker; the GS25 and GB25 are its 2.5" midranges. A shorting ring is a metal ring in the motor that lowers inductance, helping the driver play higher notes. He describes the GS8ND2 as a woofer, not a midrange. It is about 3" (76 mm) deep, made for 6×9 or 8" door openings and very small boxes: about 7 L (liters) sealed or 14 L vented, or no box at all (IB), as under seats or in Lexus rear decks. The shorting ring lets it play up to about 300 Hz, so cross it there and add a GS25 or GB25 above it. It has two 2 Ω (ohm) voice coils. Wired to 4 Ω, one woofer suits a bridged pair of 150–200 W amplifier channels ("bridged"

means two channels combined into one stronger one). A pair wired to 2 Ω suits a 500–750 W sub amp. The tag records a change: in 2016 Audiofrog said it would make no 8" and offered the GS690 6×9 instead; the GS8ND2 came in 2020, presented as a woofer.

2.1 #16 Without a sub a 6×9 makes more bass; with a sub, the GB60 is the better driver.

> The GB60 has a 2" coil, neodymium motor, shorting ring and more than twice the GS690's excursion; in a no-sub car the GS690 6×9 gives more bass.
>
> **Sources:** DIYMA 2021-08-08 · DIYMA 2020-08-25

In plain English: A 6×9 is an oval speaker, 6 by 9 inches, with a lot of cone area. Without a subwoofer, the GS690 6×9 makes more bass. With a sub handling the deep notes, the GB60 is the better driver. It has a 2" voice coil, a neodymium (strong, lightweight) magnet motor, a shorting ring, and more than twice the GS690's excursion (how far its cone can move).

2.1 #17 Under-seat midbass: keep an octave of usable bandwidth around each crossover.

> RARE In a BMW build he crossed the factory under-seat woofers at 200 Hz (as low as the 4" door speakers could play loud), so they ran about 80–200 Hz with the sub below 60 Hz. BMW under-seat boxes behave like aperiodic enclosures, borrowing frame airspace through small openings; high-Q drivers crammed in never sounded good to him.
>
> **Sources:** DIYMA 2021-08-04 · DIYMA 2020-06-24

In plain English: "Usable bandwidth" is the range of notes a speaker plays well. His rule is to leave about an octave (a doubling of frequency) of good performance on each side of every crossover. In a BMW build he crossed the factory under-seat woofers at 200 Hz, because that was as low as the 4" door speakers could play loudly. The under-seat woofers then covered about 80–200 Hz, with the sub below 60 Hz. He says BMW's under-seat boxes behave like "aperiodic" enclosures. An aperiodic box is deliberately not airtight: it breathes through damped openings, so its air is not a tight spring. These boxes borrow air from the seat frame through small openings. High-Q drivers squeezed into them never sounded good to him.

2.1 #18 Factory-door fit is set by mounting depth and the factory grille.

> GB40: 41 mm (1-5/8"), fits late BMW doors; GB60: under 2-3/4"; GS8ND2: just over 3". About 85% of G62 coaxes go under factory grilles, so it ships without one; coax tweeters sit low (GS62 2 mm, GS42 3.75 mm above the surround), and kits add multi-fit adapters and gaskets (GS682 removable gasket, GS693 six strips, GS25 foam gaskets). (Specs COMPANY.)
>
> **Sources:** Crutchfield, Audiofrog packages · CMA Networks 2023 · GS series store collection

In plain English: Mounting depth is how far a speaker sticks out behind the surface it is screwed to. Many factory doors have little room. The depths given: GB40 41 mm (1-5/8"), which fits late-model BMW doors; GB60 under 2-3/4" (about 70 mm); GS8ND2 just over 3" (about 76 mm). A coaxial speaker ("coax") has its tweeter mounted in front of the woofer. About 85% of G62 coaxes fit under the car's own grille, so it ships without one. Coax tweeters sit low to clear grilles: 2 mm above the surround on the GS62, 3.75 mm on the GS42. The kits add adapters that fit many cars and gaskets (seals): a removable gasket on the GS682, six strips with the GS693, foam gaskets with the GS25. "(Specs COMPANY.)" means these specs come from Audiofrog's unbylined company text, not from a post signed by him.

2.1 #19 Leave clearance beyond Xmax: a surround hitting the panel or grille is an install error.

> RARE Xmax is only the linear part; the cone travels further through the top-plate thickness and on momentum (GB10 10": 17.6 mm rated plus about 8 mm), so keep grilles about 1.25 in from a woofer's front. His GB60 in a gutted factory Dynaudio frame hit the Volvo's door panel. A 4" that distorts at 150–400 Hz usually has a loose mount or a surround hitting the grille.
>
> **Sources:** DIYMA 2021-12-22 · DIYMA 2021-09-22 · OFCAP Ep. 38 @33:02 · DIYMA 2021-12-02

In plain English: Xmax is how far a speaker's cone can move while staying clean (linear). He warns that the cone can travel further than that. The coil can move beyond the thickness of the magnet's top plate, and the cone can carry on moving on its own momentum. For Audiofrog's GB10 10" sub, that is 17.6 mm rated plus about 8 mm more. So keep a grille about 1.25 in (about 32 mm) from a woofer's front. He calls a surround hitting the panel or grille an installation error. It happened to him: his GB60, mounted in a gutted factory Dynaudio frame, hit the Volvo's door panel. If a 4" distorts at 150–400 Hz, the usual cause is a loose mount or a surround hitting the grille.

2.1 #20 Grilles are fine; just don't bury a mid in a tunnel.

> RARE Home speakers are designed with their grilles on; for mids a grille over about 50% open area makes no difference, and Audiofrog tweeters are designed for theirs. Listen for cloth slapping the mesh under bass, never put a midrange at the end of a 6" tunnel, expose as much open area as possible and leave real space between cone and grille.
>
> **Sources:** DIYMA 2012-08-12 · DIYMA 2015-02-27 · OFCAP Ep. 38 @33:02

In plain English: Many people remove grilles, believing they spoil the sound. He says home speakers are designed with their grilles on. For midranges, a grille that is more than about 50% open makes no difference, and Audiofrog tweeters are designed to play through their own grilles. His cautions: listen for grille cloth slapping against the metal mesh when bass plays, never put a midrange at the end of a 6" (150 mm) tunnel, expose as much open area as you can, and leave real space between the cone and the grille.

2.1 #21 Aiming a door 6" is pointless between 80 and 200 Hz, and door mounting doesn't cost depth.

> It radiates everywhere there; the real problems are path-length differences and the sub-to-midbass blend, and a small mid above it removes any need to aim. Speakers placed further forward move where the stage sits, not its depth (he says IASCA judges confuse the two).
>
> **Sources:** DIYMA 2010-02-14 · DIYMA 2013-04-27

In plain English: Between 80 and 200 Hz, a 6" speaker sends sound in every direction, so pointing it at you changes nothing. He says the real problems at those notes are path-length differences (the left and right speakers are different distances from you) and getting the sub and midbass to blend. A small midrange above the 6" removes any need to aim it at all. Many people believe door speakers can't give a deep soundstage. He says placing speakers further forward changes where the stage sits, not its depth (the sense of instruments layered from front to back). He says judges in IASCA, a car-audio competition, confuse the two.

2.1 #22 Measure before rebuilding a door location.

> Measure near-field at the grille, then at the seat: lows won't differ, highs will, which separates a location problem from a cabin problem. A flat +6 dB above 250 Hz with the trim off is the car's door-open volume cut (check the door switch), not the panel. Near-field readings can't show panel vibration or cone breakup; those appear only beyond about 3× the driver's largest dimension.
>
> **Sources:** DIYMA 2010-02-18 · DIYMA 2022-01-30 · DIYMA 2022-03-16

In plain English: "Near-field" means measuring with the microphone very close to the speaker, so it picks up mostly the speaker itself. Before rebuilding a door location, he says, measure near-field at the grille, then measure at your seat. Low notes won't differ between the two; high notes will. The comparison shows whether a problem comes from the speaker's spot or from the cabin; the notes don't spell out the comparison further. If the response is a flat 6 dB higher above 250 Hz with the door trim removed, the trim is not the cause. Some cars turn the volume down about 6 dB when they think a door is open, so check the door switch. Near-field readings also can't show panel vibration or "cone breakup" (the cone flexing instead of moving as one piece). Those appear only beyond about three times the driver's largest dimension, roughly 20 in (50 cm) for a 6.5" speaker.

2.2 Sound deadening, rattles and road noise

What this part is about: What sound-deadening material is really for, and what it can't do. He treats deadening as a way to stop rattles and panel vibration, not as a way to soak up sound or change the car's acoustics. Road noise gets its own short recipe.

2.2 #1 Deaden to stop rattles and panel movement, in a set order.

> RARE Door priority: (1) the trim panel, the worst rattle source (a large passenger leaning on it shows it); (2) the inner skin; (3) the outer skin only with material left over. Foam anything that might rattle; wrap or tie down harnesses. In his C30 he damped doors and floor, not the headliner, with Ground Zero material, no buckshot.
>
> **Sources:** OFCAP Ep. 38 @1:13:01

In plain English: Sound deadener is sheet material stuck to panels to stop them vibrating. A door has a plastic trim panel on the inside, then two metal "skins": the inner skin (facing the cabin) and the outer skin (the outside of the car). He says the goal is to stop rattles and panel movement, and gives this order for doors. First the trim panel, the worst rattle source: a large passenger leaning on it makes a loose panel obvious. Then the inner skin. Then the outer skin, only if you have material left over. Put foam on anything that might rattle, and wrap wiring harnesses in foam or tie them down. In his Volvo C30 he damped the doors and floor but not the headliner (the roof lining). He used Ground Zero material and no buckshot (lead pellets some builders pour in to add weight).

2.2 #2 Know what each material does: foam stops rattles, mass lowers panel resonance, and absorbing low frequencies takes feet of material.

> RARE Absorbers start working at about a quarter wavelength, so a 200 Hz reflection needs about 1.5 ft: padding a console is wasted. Real absorbers are membranes (MLV's job in walls) and fiber, where friction makes heat. A "deflex" pad or Dynamat just adds mass, lowering panel resonance;

> foam absorbs almost nothing.
>
> **Sources:** DIYMA 2011-11-24 · DIYMA 2015-11-15 · DIYMA 2022-01-14

In plain English: He says each material does a different job. Foam stops rattles. Mass, such as a "deflex" pad (a heavy pad stuck behind a door speaker) or Dynamat, just makes a panel heavier, which lowers the note it vibrates at (its resonance). Foam absorbs almost no sound. Real absorbers are membranes (that is MLV's job in walls; MLV is mass-loaded vinyl, a heavy flexible sheet) and fiber, where friction between the fibers turns sound into heat. Absorbers only start working once they are about a quarter of a wavelength thick, and low notes have long waves. So absorbing a 200 Hz reflection takes about 1.5 ft (45 cm) of material. Many people pad a center console to tame reflections; he says that is wasted effort.

2.2 #3 A 250 Hz peak is usually a reflection to EQ, not a panel to deaden: prove it with a tone.

> RARE A true resonance is mass and springiness trading energy (a door skin); a cabin peak at 250 Hz is usually a dimension reinforcing by reflection. Play a 250 Hz tone: if nothing buzzes, it's constructive interference, so cut it with EQ.
>
> **Sources:** DIYMA 2021-09-07

In plain English: A "peak" is a note that measures louder than its neighbors. He says people often blame a peak near 250 Hz on a vibrating panel and reach for more deadener. A true resonance is a mass and a springiness passing energy back and forth at one note, as a door skin does. A cabin peak at 250 Hz is usually something else: one of the car's dimensions makes a reflection arrive in step with the direct sound, so the two add up. His test is to play a 250 Hz tone. If nothing buzzes, it is constructive interference (two sound waves adding together), not a panel, so cut it with EQ.

2.2 #4 Road noise: deaden, then buy quiet tires, then make per-road presets; never EQ for noise above 300 Hz.

> RARE "The best midbass upgrade I ever made" was swapping aggressive Goodyears for Continentals. Record noise on your usual roads and build a couple of DSP presets (in 2008 he kept one for a coarse-pavement highway).
>
> **Sources:** DIYMA 2013-04-25

In plain English: His order for dealing with road noise: first deaden, then buy quieter tires, then make processor settings for particular roads. He calls swapping aggressive Goodyear tires for Continentals "The best midbass upgrade I ever made". For the last step, record the noise on roads you usually drive. Then build a couple of DSP presets: saved settings in your digital signal processor that you switch between. Back in 2008 he kept one for a highway with coarse pavement. His limit: never use EQ to fight noise above 300 Hz. The source doesn't spell out why he chose that limit.

2.2 #5 Don't treat the cabin to kill reflections: mats, films and carpet barely work, and it's whack-a-mole.

> RARE Fibrous dash mats absorb only the top octaves; a film neither absorbs nor diffuses (the only two ways to reduce a reflection). Chasing each reflection with shapes and foam is "like pulling weeds in the jungle". To identify one, read the impulse response: delay after the first arrival × 1,132

> ft/s gives the extra path (in a Mini, 4.9 ms was the windshield).
>
> **Sources:** DIYMA 2013-09-14 · DIYMA 2010-05-10

In plain English: A reflection is sound that bounces off a surface, such as glass or the dash, and reaches you a little after the direct sound. He says there are only two ways to reduce one: absorb it or diffuse (scatter) it. Fibrous dash mats absorb only the top octaves (the highest notes), and a film on glass does neither. Many people try to tame each reflection with shapes, foam and carpet; he calls it "like pulling weeds in the jungle". To find which surface causes a reflection, read the impulse response. That is a measurement over time that shows the direct sound arriving first, then each echo. Take an echo's delay after the first arrival and multiply by 1,132 ft/s, the speed of sound (about 345 m/s). That gives how much further the echo traveled; for example, 1 ms (a thousandth of a second) is about 1.1 ft. In a Mini, the 4.9 ms peak was the windshield.

2.2 #6 Heavy absorption sounds accurate but small.

> RARE The most absorptive car he heard (thick panels over almost all glass, speakers low or in under-dash horns) was loud, dynamic and rattle-free, like great headphones in front of you, but lacked space; he saw it only as a base for added multichannel ambience.
>
> **Sources:** DIYMA 2013-09-15

In plain English: Many people believe the fewer reflections a car has, the better it sounds. The most absorptive car he ever heard had thick panels over almost all the glass and speakers mounted low or in horns under the dash. He found it loud and dynamic with no rattles, like great headphones but with the sound in front of you. But it lacked a sense of space. He saw it only as a starting point for adding multichannel ambience, meaning extra channels and processing that recreate the feeling of a larger space.

2.2 #7 Rattles pull the bass image backward.

> RARE For bass to seem frontal, the sub must not make distortion or rattles (and must be in phase with the fronts at the crossover), so rattle control is an imaging fix. His own IB sub hangs from the package shelf on rubber isolation mounts to avoid rattles.
>
> **Sources:** Tuning PDF v1 (~2018) · DIYMA 2017-12-09

In plain English: "Imaging" is where you hear each sound coming from. A subwoofer usually sits behind you, in the trunk or rear deck, yet bass should seem to come from the front. He says that only happens if the sub makes no distortion or rattles, and is in phase with the front speakers at the crossover (both pushing air together there). Rattles therefore pull the bass toward the back, so stopping them is part of fixing the image. His own IB sub hangs from the package shelf (the shelf behind the rear seat) on rubber isolation mounts, to avoid rattles.

2.2 #8 Build sub boxes rigid and airtight; beyond that the box matters only near resonance.

> A couple of octaves above resonance a box has little effect unless it "is leaky, flexes a lot or buzzes". Line it with about 1" of fiberglass or dacron and model the stuffing (he used LEAP); polyfill mainly absorbs high frequencies far above a sub's range.
>
> **Sources:** DIYMA 2008-05-26 · DIYMA 2013-10-30

In plain English: His basic rule for subwoofer boxes is to make them stiff and airtight. Beyond that, he says, the box only shapes the sound near the speaker's resonance. A couple of octaves above it (for example, from about 160 Hz up if the resonance is 40 Hz), a box has little effect unless it "is leaky, flexes a lot or buzzes". He says to line the box with about 1" (25 mm) of fiberglass or dacron (polyester fiber) and to include the stuffing when you model the box in design software; he used LEAP. Polyfill (pillow-type stuffing) mainly absorbs high notes, far above the range a sub plays.

2.2 #9 Prove what treatment does by recording it.

> RARE Audiofrog suggests binaural recordings of treated and untreated cars to show treatment's effect, and recording a car's ambient noise to set "masking equalization". (COMPANY.)
>
> **Sources:** Audiofrog Binaural page

In plain English: A binaural recording is made with two microphones in the ears of a dummy head, so that playback on headphones sounds as if you were there. Audiofrog's company page suggests two uses. First, record treated and untreated cars this way to show what sound treatment actually changes. Second, record a car's background noise to set "masking equalization". Masking means a steady noise makes quieter sounds of similar pitch harder to hear; the page doesn't explain the method further. "(COMPANY.)" marks this as unsigned Audiofrog company text, not a post by him.

2.2 #10 Not found in anything he said:

> CLD coverage percentages, MLV as a car noise barrier, closed-cell decoupler layers, door water management, a step-by-step rattle hunt, or brand-versus-brand deadener comparisons. The OFCAP 38 order above is his only stated deadening procedure.

In plain English: This is a researchers' note, not one of his claims. They searched his material and found nothing on these topics. CLD coverage percentages: how much of a panel to cover with constrained-layer damping, the foil-backed sheets such as Dynamat. MLV as a noise barrier in cars. Closed-cell "decoupler" foam layers. Managing water inside doors. A step-by-step hunt for rattles. Brand-versus-brand deadener comparisons. The order he gave on the OFCAP podcast, Episode 38 (point 2.2 #1 above), is the only deadening procedure he stated. So advice on those other topics should not be credited to him.

2.3 Where midranges and tweeters go

What this part is about: Where to put midranges (mid-size speakers for voices and most instruments) and tweeters (small speakers for the highest notes). His main idea is that the windshield, dash and side windows bounce sound back. Those bounces cancel certain notes, so a speaker should sit either right in a corner or surface, or well away from it, never hanging a short distance off it. The part ends with the products and the cars he built.

2.3 #1 A-pillars are "absolutely the worst places to mount speakers", especially mids, and delay can't rescue them.

> RARE A mid a few inches from the windshield/dash/side-glass apex makes each surface a second source with its own comb filter; you can't delay a reflection apart from its source. A pod about a foot off that corner puts a 2-ft reflection path and a massive hole near 250 Hz. The only gain is

> height, which a correct crossover gives anyway.
>
> **Sources:** DIYMA 2012-09-03 · DIYMA 2013-11-20 · DIYMA 2021-08-26

In plain English: An A-pillar is the slanted post on each side of the windshield. Many people put mids in A-pillar pods to lift the soundstage. A comb filter is what happens when a sound mixes with a slightly late copy of itself, such as a reflection: some notes cancel and others add, in a regular pattern like the teeth of a comb. He says a mid a few inches from the corner where windshield, dash and side glass meet turns each of those surfaces into a second source, each with its own comb filter. Delay (time alignment) can't fix this: a reflection is made by the speaker's own sound, so you can't delay one apart from the other. A pod about a foot (30 cm) off that corner gives a 2-ft (60 cm) reflection path and a massive hole near 250 Hz. The pillar's only gain is a higher stage, which he says a correct crossover gives anyway.

2.3 #2 If you must use the pillar, push the pod into the base of the corner, flush, and never fire mids up at the glass.

> RARE Flush in the pillar firing across the dash is fine with a low enough crossover, and closer to the corner is better. Mids fired at the windshield or ceiling become omnidirectional horizontally, smearing toward the near side glass: "the most dense comb filter possible".
>
> **Sources:** DIYMA 2021-09-07

In plain English: "Flush" means mounted so the speaker's face is level with the surface instead of sticking out. If you must use the pillar, he says, push the pod into the base of the corner and mount it flush. Flush in the pillar, firing across the dash, is fine if the crossover is low enough (the source gives no number), and the closer to the corner the better. Never fire mids up at the windshield or ceiling. Aimed that way, at midrange notes they spread sound in every horizontal direction. The image then smears toward the side window nearest you, and he calls the result "the most dense comb filter possible".

2.3 #3 Use a tape measure before building: distance to the nearest surface sets where the reflection dip lands.

> RARE A null falls where the reflected path is half a wavelength longer than the direct one: a mid 6" from a surface (about 1 ft extra path) dips near 500–550 Hz, repeating near 1.5, 2.5, 3.5 kHz; 1 ft away, near 280 Hz. Nulls can't be EQ'd and smear imaging. To find a culprit, match the null to a surface's extra path and block it to confirm.
>
> **Sources:** Audiofrog blog Jun 2021 · DIYMA 2020-10-07 · DIYMA 2021-11-06

In plain English: A null is a deep dip where sound cancels. A reflection travels further than the direct sound. At the note where that extra distance is exactly half a wavelength, the two arrive opposite, one pushing as the other pulls, and cancel. So the distance from a speaker to the nearest surface decides where its dip lands, and he says to check it with a tape measure before building. His numbers: a mid 6" (15 cm) from a surface has about 1 ft (30 cm) of extra path and dips near 500–550 Hz, repeating near 1.5, 2.5 and 3.5 kHz. A mid 1 ft away dips near 280 Hz. He says nulls can't be fixed with EQ, and they smear the imaging. (Boosting a note boosts its reflection too, so the two still cancel.) To find which surface is to blame, match the null's frequency to a surface whose extra path fits, then block that path to confirm.

2.3 #4 Tuck mids into a corner or flush in a surface; keep tweeters away from surfaces.

> RARE Close to a surface pushes the null above the mid's low-pass; mounting in the surface removes it (why dash corners work great). Pillars fail because they are near several surfaces but flush in none: spreading chaos isn't reducing it. Position relative to glass and dash matters more than mid-to-tweeter spacing, which forums obsess over.
>
> **Sources:** DIYMA 2013-08-10 · DIYMA 2021-12-24

In plain English: This follows from the point above: the closer a speaker is to a surface, the higher its dip. He says to tuck a mid very close to a surface so the dip moves above its low-pass, the top of the range the mid plays, where it no longer matters. Mounting the mid in the surface (flush) removes the dip, because that surface becomes part of the baffle instead of a reflector. That is why dash corners work so well for mids. Tweeters go the other way: keep them away from surfaces, so the dip falls below the notes they play. Pillars fail because they are near several surfaces but flush in none; spreading the problem around is not the same as reducing it. Many people on the forums obsess over the distance between mid and tweeter. He says the mid's position relative to the glass and dash matters more.

2.3 #5 Best front layout: tweeter in the sail (mirror) panel, mid at the top of the door below it.

> RARE It moves both away from the windshield and makes door and window the baffle (bigger baffles mean fewer nearby reflections). A sail tweeter about 1.5 in from the glass dips near 2.2 kHz, low enough not to matter, and gives a wider stage than dash or pillar.
>
> **Sources:** DIYMA 2011-11-05 · DIYMA 2017-10-31

In plain English: The sail panel is the small triangular panel at the front corner of the front door window, where the mirror usually mounts. His preferred layout is the tweeter in the sail panel and the mid at the top of the door, just below it. That moves both away from the windshield, and it makes the door and window act as the baffle. A bigger baffle means fewer nearby surfaces to reflect from. A sail tweeter about 1.5 in (38 mm) from the glass has its reflection dip near 2.2 kHz, which he says is low enough not to matter. He also finds sail tweeters give a wider stage than dash or pillar ones.

2.3 #6 Stack the tweeter directly above the mid, as home speakers do.

> RARE Vertical stacking keeps mid/tweeter alignment at every horizontal angle (only vertical angles cause lobing); side-by-side works much worse. With good crossovers you hear each pair at the higher driver, so width doesn't come from the mids unless the tweeter is crossed absurdly high (about 7 kHz). His own doors are built this way.
>
> **Sources:** DIYMA 2021-09-07

In plain English: Home speakers stack their drivers in a vertical line, and he says car speakers should too: the tweeter directly above the mid. Stacked that way, the mid and tweeter stay in step at every left-to-right angle. Only up-and-down angles cause "lobing", where the two drivers partly cancel each other near the crossover because their distances to you differ. Side by side works much worse. He explains that with good crossovers you hear each pair of drivers as coming from the higher-pitched one, so a mid/tweeter pair sounds as if it comes from the tweeter. So stage width doesn't come from the mids, unless the tweeter is crossed absurdly high (about 7 kHz). His own doors are built this way.

2.3 #7 Kick panels: better geometry, not a cure.

> RARE They cut the left/right path difference from about 28" (center cancelling near 250 Hz) to about 6", "Better, but not fixed", and sit far from the windshield corner. But the footwell often makes a big low-midrange dip (EQ can't fill it), legs block them, midbass boxes are hard to fit, and they fail when you sit high above them (over about 20° down: minivans, trucks). The passenger kick measured as the strongest midbass spot.
>
> **Sources:** Audiofrog blog Dec 2020 · DIYMA 2016-04-27 · DIYMA 2021-09-16

In plain English: Kick panels are the side panels in the front footwells, near your feet. The driver sits much closer to the left speakers than the right. With door mids the difference is about 28" (71 cm), which makes left and right cancel each other for center sounds near 250 Hz. Kick panels cut that difference to about 6" (15 cm): "Better, but not fixed". They also sit far from the windshield corner. The downsides he lists: the footwell often causes a big dip in the low midrange that EQ can't fill, your legs block the sound, and midbass boxes are hard to fit there. They also fail when you sit high above them, more than about 20° downward, as in minivans and trucks. In his measurements, the passenger kick panel was the strongest spot for midbass.

2.3 #8 Dash mounts get big free low-end gain: cut it, don't boost the "missing" top.

> RARE The dash cuts the radiating space by 3/4 or more (+6 dB per halving) and the windshield adds more, so a 2.5" in the dash can play down to 200–250 Hz with care (OEMs give it only 15–20 W). The apparent high-frequency roll-off is really a low-frequency hump: cutting it, even 12 dB, just saves power and excursion.
>
> **Sources:** Audiofrog blog Jun 2021 · DIYMA 2020-08-25 · CMA Connected Oct 2021 @34:22

In plain English: This uses the rule from point 2.1 #8: about +6 dB each time the space a speaker radiates into is halved. He says a dash cuts that space by 3/4 or more (two or more halvings), and the windshield adds more on top. So a 2.5" (63 mm) dash speaker can play down to 200–250 Hz with care, even though carmakers give it only 15–20 W. On a measurement, a dash mid can look as if its high notes are fading. He says that is really a hump at the low end. Many people boost the "missing" highs. He says to cut the hump instead, even by 12 dB, which only saves amplifier power and cone movement.

2.3 #9 A small dash mid aimed at a raked windshield is a "poor man's tweeter", which is why the GS25 exists.

> RARE Its narrow treble beam bounces off the glass to everyone, so beaming stops mattering (not with upright glass, as in a Jeep). GM, Toyota, Chrysler, Bose and Harman use this "middler"; the GS25 drop-in does the job with far more power handling (high-pass 200 Hz or above, 12 dB/oct or steeper). A real tweeter is still better. **Conflict:** this uses the windshield reflection his custom door 3-way (2.6 #6) minimizes; it is a factory-location strategy.
>
> **Sources:** DIYMA 2022-01-03 · CMA Connected Oct 2021 @32:43 · GS25 page

In plain English: "Raked" means sloped back. Higher notes "beam", meaning they come out in a narrow beam rather than spreading, so you only hear them fully if the speaker points at you. He says a small mid in the dash, aimed at a sloped windshield, sends its narrow treble beam onto the glass, which bounces it to everyone in the car, so beaming stops mattering. It doesn't work with upright glass, as in a Jeep. GM, Toyota, Chrysler, Bose and Harman use this small dash speaker, the

"middler", and he calls it a "poor man's tweeter". Audiofrog's GS25 drops into that spot and handles far more power. Use a high-pass at 200 Hz or above, at 12 dB per octave or steeper (the slope says how fast lower notes are cut). He adds that a real tweeter is still better. The [Conflict] tag notes the tension in his advice. This trick relies on the windshield reflection, which his custom door 3-way design (point 2.6 #6) tries to keep small. The tag files it as a strategy for factory speaker locations.

2.3 #10 For a dash mid under a factory grille, put a small tweeter under the same grille rather than adding pods.

> RARE Clear the plastic with a rotary tool; this works if the mid still responds to EQ up to about 4 kHz. Keep the tweeter base off a 5/8 in gap to the windshield (big dip near 10 kHz), and don't leave a mid 3/4 in behind a grille piece: mount it flush or cut the plastic out and use mesh plus cloth.
>
> **Sources:** DIYMA 2021-09-05

In plain English: Many people add separate tweeter pods on the dash. For a dash mid under the factory grille, he suggests putting a small tweeter under the same grille instead. Clear away the plastic behind the grille with a rotary tool (a small high-speed cutting tool) and mount the tweeter there. This works if the mid still responds to EQ up to about 4 kHz, so the tweeter only has to fill in the very top. Avoid placing the tweeter's base 5/8 in (16 mm) from the windshield: that gap causes a big dip near 10 kHz. And don't leave a mid sitting 3/4 in (19 mm) behind a piece of grille plastic. Either mount it flush, or cut the plastic out and use mesh covered with grille cloth.

2.3 #11 A top-of-dash center: once a "frequency response nightmare", later the place to be.

> Firing at the glass, a dash L/C/R gives "three left speakers, three right speakers and two center channels" and confuses auto time alignment (try reversing the center's polarity); make sure the baffle doesn't leak. By 2021 he advised mounting it as close to the windshield–dash junction as possible: its broad dip near 2 kHz mimics the dip a phantom center has, so it sounds familiar. [Refinement, not reversal: the 2010 "nightmare" remark was about response ("though it does raise the stage"), and the MS-8 FAQ (2011 or earlier) already named a big dash mid firing at the windshield his preferred center ("This works great"; 1.10 #5); 2021 adds placement at the junction and the 2 kHz rationale from Shively and House AES papers.]
>
> **Sources:** DIYMA 2010-08-15 · DIYMA 2021-09-22

In plain English: A center channel is a mid-dash speaker for centered sounds, such as a singer; "L/C/R" means left, center and right. In 2010 he called a top-of-dash center firing at the glass a "frequency response nightmare". With all three dash speakers bouncing off the glass, you get "three left speakers, three right speakers and two center channels", and that confuses automatic time alignment (processors that set delays by themselves). His tip was to try reversing the center's polarity, and to make sure its baffle doesn't leak. By 2021 he advised mounting it as close as possible to where the windshield meets the dash. The reflection there gives a broad dip near 2 kHz. A "phantom center", the center image you hear between two stereo speakers, also has a dip near 2 kHz, so the real center sounds familiar. The tag explains this is a refinement, not a change of mind. The 2010 remark was about frequency response, and even then he said it "does raise the stage". His MS-8 FAQ (2011 or earlier), written about the JBL processor he worked on at Harman, already named a big dash mid firing at the windshield as his preferred center ("This works great"). The 2021 advice adds the exact spot and the 2 kHz reasoning, taken from audio-engineering (AES) papers by Shively and House.

2.3 #12 Every location has its dip; choose which one you can live with.

> RARE Mids low in his doors accept a side-glass dip for width; all-in-the-dash loses width and adds a windshield dip; kicks add a midbass dip ("It's always something"). For height, move the mid to the door top; for width plus height, everything in the doors and a center in place of a dash gauge pod. A 300–500 Hz dip is usually the door window; don't hunt endlessly.
>
> **Sources:** DIYMA 2013-08-31 · DIYMA 2021-11-02

In plain English: He says there is no perfect speaker location; each one has its own dip, so choose the one you can live with. His own car has mids low in the doors, which accepts a dip caused by the side glass in exchange for stage width. Putting everything in the dash loses width and adds a windshield dip. Kick panels add a midbass dip. "It's always something." If you want more height, move the mid to the top of the door. For width plus height, put everything in the doors and add a center speaker where a dash gauge pod (a housing for extra dials) would sit. A dip at 300–500 Hz is usually the door window reflecting, and he says not to hunt for it endlessly.

2.3 #13 Map candidate locations with a speaker at the driver's head and a mic at each spot (acoustic reciprocity).

> RARE Mic and speaker swap roles where dispersion is wide, so it works for sub, midbass and mid spots but not tweeters. His rig: a GB25 in a 3D-printed pod at head height and a 3D-printed boundary mic at dash top, pillar, kick, door tops and trunk. From 2023 he taught "system design" this way rather than guessing locations.
>
> **Sources:** OFCAP Ep. 38 @36:00 · Absolute Electronix 2023 @1:00:14

In plain English: Acoustic reciprocity is a physics principle: the sound path from point A to point B behaves the same as from B to A, so a speaker and a microphone can swap places. He uses it to test spots before cutting anything. Put a small speaker where the driver's head is and a mic at each place you might mount a speaker, then compare the measurements. It only holds where the speaker spreads sound widely, as a mic hears from all directions. So it works for sub, midbass and mid locations, but not tweeters, which beam. His rig: a GB25 in a 3D-printed pod at head height, and a 3D-printed boundary mic (a mic made to sit flat on a surface) placed on the dash top, pillar, kick panel, door tops and trunk. From 2023 he taught "system design" this way instead of guessing locations.

2.3 #14 Which Audiofrog set for which location.

> GB25 2-way: A-pillars or center; GB15 + GB60: "the primo set" for kick panels (the GB15 is designed for the GB60); GB10 + GB25 + GB60: the ultimate active system for pillars with door or kick midbass; GB10/GB15 + GB40: doors (BMW) or center; GB25 in any 3-way when only a small mid fits; G62 coax at the bottom of the door or in the rear deck. (COMPANY and retailer.)
>
> **Sources:** Crutchfield, Audiofrog packages · GB25 page · GB15 page

In plain English: This is a product guide from Audiofrog and a retailer (Crutchfield), not his signed posts; that is what "(COMPANY and retailer)" means. In Audiofrog's naming, the GB10 and GB15 are tweeters (1" and 1.5"), the GB25 is a 2.5" mid, the GB40 is a 4" mid, the GB60 is a 6" (165 mm) midbass, and the G62 is a coaxial speaker. The pairings: a GB25 two-way for A-pillars or a center. GB15 plus GB60, "the primo set", for kick panels; the GB15 was designed to go with the GB60. GB10 plus GB25 plus GB60 as the ultimate "active" system, where each driver has its own amplifier

channel and processor crossover, for pillars with midbass in the doors or kicks. GB10 or GB15 plus GB40 for doors (as in BMWs) or a center. A GB25 in any three-way where only a small mid fits. The G62 coax at the bottom of the door or in the rear deck.

2.3 #15 His Harman-era Volvo S60 (c. 2008–10) shows the priorities.

> RARE Dual-coil 6" midbass at the front of the factory lower grille, nothing in the factory 4" spot behind it ("a stupid place for a midrange"), a 3" mid ahead of the door handle, a 1" tweeter in a sail he built, a 6" plus tweeter in the dash center, two 10" IB subs in the deck. He ranked center channel, door-mid location, lots of EQ and center steering highest, and praised cars built for audio (big door openings, trunk battery, deck room).
>
> **Sources:** DIYMA 2010-02-19 · DIYMA 2009-05-27

In plain English: This describes his own car from about 2008–10, when he worked at Harman, the company behind JBL and Infinity. He put dual-coil 6" midbass speakers at the front of the factory lower grille, and nothing in the factory 4" spot behind it ("a stupid place for a midrange"). A 3" mid went ahead of the door handle, and a 1" tweeter in a sail panel he built. In the middle of the dash he put a 6" plus a tweeter as a center channel, and two 10" IB subs went in the rear deck. He ranked the center channel, the door-mid location, lots of EQ and center steering highest. Center steering is processing that sends center sounds to the center speaker. He also praised cars built with audio in mind: big door openings, the battery in the trunk, and room in the rear deck.

2.3 #16 His Audiofrog demo cars use the same recipe.

> Mercedes (CES 2016): identical active GB10 + GB25 + GB60 3-ways in all doors and sides, GB10 + GB40 deck pairs, a machined-aluminum center pod, a custom 24-channel amp. Volvo C30 (2023, 7.1, 18 speakers): door 3-ways, a GB60 + GB10 center so it can play loud (2" coil), two GB10D4 subs in a small sealed box, a head unit that kept the factory HVAC display so the dash stayed uncut, timed turn-on sequencing.
>
> **Sources:** Ricardo Rangel, CES 2016 · OFCAP Ep. 38 @9:02 · Absolute Electronix 2023 @51:00

In plain English: Audiofrog's demonstration cars follow the same plan as his own. The Mercedes, shown at CES 2016 (a large electronics trade show), had identical active three-ways in all the doors and sides: GB10 tweeter, GB25 mid and GB60 midbass, each on its own amplifier channel. It also had GB10 plus GB40 pairs in the rear deck, a center pod machined from aluminum, and a custom 24-channel amplifier. The Volvo C30 (2023) is a 7.1 system (seven main channels plus one bass channel) with 18 speakers. It has three-ways in the doors and a GB60 plus GB10 center, so the center can play loud (the GB60 has a 2" voice coil). Two GB10D4 10" subs sit in a small sealed box. Its head unit (the radio and source unit) kept the factory climate-control (HVAC) display, so the dash stayed uncut. The equipment also switches on in a timed sequence.

2.4 Pods, back volumes and fabrication

What this part is about: A "pod" is a small custom housing that holds a speaker somewhere the car didn't provide a spot, often on the windshield pillars. This part covers how much air a speaker needs behind it, how to seal and test a pod, practical building methods (grilles, 3D printing, upholstery, cutting holes), and how much custom fabrication is actually worth doing.

2.4 #1 A pod is an enclosure: size the box to the driver, not the biggest driver to the pillar.

> RARE An IB-type driver crammed into a pod no bigger than its basket becomes a high-Q mid with no bass and an untamable peak, and a "distortion machine" at its raised resonance, so people cross it low and make it worse. "Put the biggest speaker possible up there" is too simple to be useful.
>
> **Sources:** DIYMA 2013-12-14 · DIYMA 2017-10-01 · DIYMA 2021-11-19

In plain English: Some background first. A speaker's moving part, the cone, has a natural "resonance": a frequency where it vibrates most easily. Air sealed in a small box acts like a spring on the cone and pushes that resonance up. "Q" describes how peaky the response is around resonance; high Q means a sharper hump. An "IB" (infinite baffle) driver is one designed to work with a very large or open air space behind it, such as a door or trunk. His point is that a pod is a box, so it changes the speaker. Squeeze an IB-type driver into a pod barely bigger than its frame and it loses its bass, gains a peak you can't tame, and distorts heavily around its new, higher resonance. People then set the crossover low to get more out of it, which feeds it exactly the frequencies where it is worst. Many people believe "put the biggest speaker possible up there"; he says that advice is too simple to be useful. Pick a driver that suits the box you can actually fit.

2.4 #2 High-passed at about 200 Hz LR4, a small mid barely cares what box it is in; open-back is often best.

> RARE Modeled GB25 responses in IB, 0.25 L and the smallest box that covers its back (about 0.162 L) were nearly identical, so venting into the pillar is optional. Its 2015 brief: reach 200 Hz in 0.004 ft³ while taking 100 W RMS / 300 W peak, because "0.5 cubic foot pods look ridiculous on A-pillars". The GB40 wants IB or at least 1 L. A mid crossed above resonance is usually better with no box at all (Fs is always below Fc), given a big enough baffle.
>
> **Sources:** DIYMA 2017-07-02 · DIYMA 2015-05-08 · DIYMA 2021-11-19

In plain English: A high-pass filter lets only frequencies above a set point reach a speaker. "LR4" (Linkwitz-Riley, 4th order) is a common steep filter that cuts 24 dB per octave. The GB25 is Audiofrog's 2.5" (63 mm) midrange; the GB40 is a 4". A liter is roughly a quart, and 0.004 ft³ is about 0.11 liters, which is tiny. His point: once a small mid only plays above about 200 Hz, the box hardly matters. His models of the GB25 with no box (IB), in 0.25 L, and in the smallest box that covers its back (about 0.162 L) came out nearly identical, so letting it breathe into the pillar is optional. Its design brief, as he described it in 2015, was to reach 200 Hz in that tiny space while handling 100 W continuous and 300 W peaks, because pods of 0.5 cubic foot (about 14 liters) look ridiculous on A-pillars. The larger GB40 needs IB or at least 1 liter. The bracket note means: a speaker's resonance with no box (Fs) is always lower than in a sealed box (Fc), so a mid crossed above resonance is usually better with no box at all, as long as the panel it sits in (the baffle) is big enough.

2.4 #3 Seal and gasket every pod: a small air leak "rattles".

> RARE A customer's GB25 buzzed only in its pod (bumps, voids, no gasket), and a respected installer blamed the speaker. Kill leaks and openings that let front and back mix, use the supplied foam gasket on a flat baffle, and mount small mids in something sturdy or damped, since they shake at high levels.
>
> **Sources:** DIYMA 2022-03-01 · DIYMA 2022-03-16

In plain English: A gasket is a soft seal, here foam, between a speaker's frame and the surface it mounts on. The front and back of a cone push out opposite sound waves; where they meet through a gap they partly cancel, and air forced through a small gap can buzz. His story: a customer's GB25 buzzed only when it was in its pod. The pod had bumps, voids and no gasket, yet a respected installer blamed the speaker. His point is that a small air leak can sound just like a rattling speaker. So in practice: close every leak or opening that lets front and back sound mix, use the foam gasket supplied with the speaker on a flat mounting surface, and mount small mids in something sturdy or damped, because they shake when played loud.

2.4 #4 Check a pod with an impedance sweep, and measure the in-pod resonance before choosing a crossover.

> RARE A leak shows as an extra, port-like resonance ("An enclosure has to actually be an enclosure"); the pod's real volume is Vb = Vas / ((Qtc/Qts)² − 1). Without a DATS, put a multimeter on current in series with the speaker and play a slow, quiet sine sweep: minimum current marks Fs (or Fc in a box).
>
> **Sources:** DIYMA 2022-03-01 · DIYMA 2021-08-26

In plain English: Impedance is how hard a speaker resists the amplifier's signal, and it changes with frequency, peaking at the speaker's resonance. An "impedance sweep" measures it across the frequency range; DATS is a popular tester that does this. His point: a leaky pod shows an extra resonance, like the vent (port) of a ported box. As he puts it, "An enclosure has to actually be an enclosure." The formula lets you work out the pod's real air volume (Vb) from two figures on the speaker's data sheet (Vas and Qts) and the Q you measure in the box (Qtc). Measure the resonance in the pod before you pick a crossover point. With no DATS, set a multimeter to measure current, wire it in series (in line) with the speaker so all the current passes through it, and play a slow, quiet sine sweep, a pure tone gliding through the frequencies. Where the current is lowest is the resonance: Fs for a bare speaker, Fc in a box. That works because impedance is highest at resonance, so the least current flows there.

2.4 #5 His recipe for a factory-looking recessed grille in a plastic door panel.

> RARE Tape the outline; drill a ¼" pilot ¼" inside it; cut from behind with a carbide burr ⅛" inside; file to the line. Clamp matching perforated plastic grille stock between two 12" MDF plates, heat with a heat gun about a foot away (edges more than center) until it sags, press in a rounded MDF plug (hole size minus grille thickness plus 1/16–1/8"), let it cool.
>
> **Sources:** DIYMA 2012-09-08

In plain English: This is his step-by-step method for adding a speaker grille to a plastic door panel so it looks factory-made. MDF is medium-density fiberboard; a carbide burr is a hard cutting bit for a rotary tool; grille stock is plastic sheet full of small holes. Step by step: (1) mark the outline of the opening with tape; (2) drill a ¼" (about 6 mm) starter hole ¼" inside that line; (3) working from the back of the panel, cut out the opening with the burr, staying ⅛" (about 3 mm) inside the line; (4) file the edge out to the line for a clean finish. For the grille itself: (5) clamp perforated plastic that matches the factory grilles between two 12" (about 300 mm) MDF plates, each with a matching hole; (6) warm it with a heat gun held about a foot (30 cm) away, heating the edges more than the

middle, until the plastic sags; (7) press in a round-edged MDF plug, sized to the hole minus the grille's thickness plus 1/16–1/8" (about 1.5–3 mm), and let it cool. The plastic sets in a dish shape, giving a recessed grille.

2.4 #6 3D-printed parts: design for the process and for cabin heat.

> RARE If a vinyl-wrapped pod can't be covered, the design is wrong. FDM can't print 1.5 mm holes on 2 mm centers; resin can. Print a grille with its ring (separate thin grilles warp). A part fitted on a 90 °F day warped 1/16–1/8" between bolts: check heat-deflection temperature (about 105 °C for "automotive" resin), blend a flexible resin with brittle high-temp resin, and use glued-in studs, not screws into bosses. PLA's ~88 °C rating is not a 3-lb speaker hanging on four screws.
>
> **Sources:** OFCAP Ep. 38 @15:08 · DIYMA 2021-07-02

In plain English: 3D printers come in two common kinds. FDM printers lay down melted plastic strands; resin printers harden liquid resin with light and can print finer detail. "Heat-deflection temperature" is the point at which a plastic starts to bend under load. PLA is a common, cheap printing plastic. "Bosses" are the raised posts that screws go into. His points: if a pod you plan to wrap in vinyl can't be covered, the design is wrong. FDM can't print 1.5 mm holes spaced 2 mm apart (center to center); resin can. Print a grille in one piece with its ring, because separate thin grilles warp. Car interiors get hot: a part fitted on a 90 °F (32 °C) day warped 1/16–1/8" (about 1.5–3 mm) between its bolts. So check the heat-deflection temperature (about 105 °C for "automotive" resin), blend a flexible resin with a brittle high-temperature one, and use glued-in studs rather than screws into printed posts. PLA's roughly 88 °C rating is for the plastic on its own; a 3 lb (about 1.4 kg) speaker hanging on four screws is a far harder job.

2.4 #7 Padding under vinyl: stretch and staple, don't glue.

> RARE For padded vinyl on MDF, pre-stretch foam-backed headliner, tack it with a few staples, then stretch the vinyl over and staple both (around the back, inside the baffle opening). Gluing to foam or carpet makes bumps. Convex shapes can't be done this way: sand smooth and glue vinyl directly.
>
> **Sources:** DIYMA 2021-08-25

In plain English: Upholsterers often put a thin layer of padding under vinyl so a panel feels soft. MDF is medium-density fiberboard; headliner is the fabric used on car ceilings, usually with a thin foam backing. His method: first stretch foam-backed headliner over the MDF and hold it with a few staples, then stretch the vinyl over it and staple both layers together, wrapping them round to the back and inside the speaker opening. Don't glue the vinyl to foam or carpet, because that makes bumps. He adds that convex (curved) shapes can't be done this way: for those, sand the part smooth and glue the vinyl straight onto it.

2.4 #8 Cut baffle holes with a router and circle jig, not a freehand jigsaw.

> "Either the speaker fits or it doesn't fit": he used it as the analogy for objective tuning.
>
> **Sources:** Mobile Electronics column Sep 2017

In plain English: A router is a spinning cutting tool. A circle jig is an arm that pivots on a center point so the router cuts a perfect circle. A jigsaw guided by hand tends to wander. His point is to cut speaker holes with a router and circle jig, because "either the speaker fits or it doesn't fit." In a

2017 magazine column he used this as a comparison for tuning: he argued for objective, measurement-based tuning, where there is a right answer you can check, just as a hole either fits or doesn't.

2.4 #9 Competent installation is simple: solid mounting, a properly designed sub box, the right locations.

> RARE Laser aiming and midbass door boxes are "complexity for complexity's sake"; dash rebuilds for symmetry, moved seats, diffusers, foam and carpet come after "everything else in life is done". What makes great systems is installers who know how to install the products.
>
> **Sources:** DIYMA 2010-02-18 · BusinessWorld 2024

In plain English: His point is that a competent installation is not complicated: mount speakers solidly, build a properly designed subwoofer box, and put speakers in the right locations. Things like aiming speakers with lasers or building boxes for the woofers inside the doors he calls "complexity for complexity's sake". Rebuilding a dashboard so both sides match, moving the seats, and adding diffusers, foam or carpet come only after "everything else in life is done," meaning they are last on the list. What makes great systems, he says, is installers who know how to install the products.

2.4 #10 Fabricate for sound, not for photos; use factory locations that work.

> RARE He fabricates only to put a speaker in the right spot. Shops show him beautiful four-month builds excused as "not bad for the 20 minutes" of tuning. His mock judging score: 1,000 points for sounding great, 25 for looks. Factory locations were chosen by PhD engineers for a reason: rearranging the car to fit "the thing we've always installed" is a big mistake.
>
> **Sources:** DIYMA 2013-11-24 · CMA Connected Oct 2021 @22:22 · CMA Connected Oct 2021 @37:25

In plain English: He builds custom parts only to get a speaker into the right spot, not to make the car look impressive. Shops show him beautiful builds that took four months, then excuse the sound with "not bad for the 20 minutes" spent tuning. If he were judging, he would give 1,000 points for sounding great and 25 for looks. He also says carmakers' speaker locations were chosen by engineers with PhDs for a reason, so tearing up a car to fit "the thing we've always installed" is a big mistake. So in practice: use factory locations that work, and put the effort into sound.

2.4 #11 Never invent on the critical path: on a paid job, do only what you already know how to do.

> RARE His C30 took about six months; quoted from scratch by a shop it would be about $15–17K of gear and $60K of labor (a show report put custom work at $20K of a $35K system). Schedule skill-building days and split a new capability (CAD, printing, design) across staff before selling it.
>
> **Sources:** OFCAP Ep. 38 @24:00 · ecoustics, T.H.E. Show 2023

In plain English: "Critical path" is a project-management term for the chain of tasks that sets the finish date. Here it means a paying customer's job. His point: on a paid job, only do what you already know how to do; never learn a new technique on a customer's car. For scale, his own Volvo C30 build took about six months. If a shop quoted it from scratch, he estimated about $15,000–17,000 of equipment and $60,000 of labor. A show report gave a different split: $20,000 of custom

work in a $35,000 system. So in practice: schedule separate days for building skills, and spread a new capability (CAD, which is computer-aided design, 3D printing, design) across several staff before you sell it.

2.5 Tweeter mounting

What this part is about: A tweeter is the small speaker that plays the highest sounds. This part covers how the way a tweeter is mounted changes its sound, whether and how far to aim it, and why certain designs and add-ons (special domes, waveguides, coaxial layouts, mounting hardware) behave as they do.

2.5 #1 Every dome tweeter has a dip set by dome height above the baffle, and the mounting moves it.

> RARE Sound travelling from the dome to the baffle and back cancels where that path is half a wavelength (near 12 kHz, depending on dome height); on a flat baffle it shows, in a pillar or sail it moves or vanishes, so baffle-mounted review curves mislead. A ring radiator (dome pinned at its center) lowers the "dome" and pushes the hole higher, with the same dispersion.
>
> **Sources:** Erin's Audio Corner GB15 · DIYMA 2014-06-11 · DIYMA 2021-12-04

In plain English: A dome tweeter's moving part is a small dome that stands up from its faceplate, and the "baffle" is the surface it is mounted in. A wavelength is the length of one sound wave; high notes have short ones (at 12 kHz, about 29 mm or just over an inch). When two copies of a sound arrive half a wavelength apart, they cancel and leave a dip. His point: some sound travels from the dome down to the baffle and back, and where that extra path equals half a wavelength (near 12 kHz, depending on how tall the dome is) it cancels. On a flat test baffle the dip shows up; mounted in a pillar or sail, the dip moves or disappears. So review graphs measured on flat baffles can mislead you about how a tweeter will behave in a car. A "ring radiator" is a design whose dome is held down at its center, so it acts like a lower dome; that pushes the dip higher, and it spreads sound the same way.

2.5 #2 The GB15 and GB10CF are shaped for car mounting, not a flat test baffle.

> RARE GB15: a tall, stiff dome in a cup whose ring is a compromise between flush (door top, kick) and unbaffled (sail, pillar). GB10CF: a lens that lengthens the path from the dome center to avoid the half-wave dip, with the stiff dome's 16 kHz peak pushed off axis where the longer in-car path softens it; high-pass 2 kHz or higher, LR4 (COMPANY).
>
> **Sources:** Erin's Audio Corner GB15 · Introducing the GB10CF

In plain English: The GB15 is Audiofrog's 1.5" (38 mm) tweeter and the GB10CF is its 1" carbon-fiber tweeter. "Off axis" means to the side of the direction a speaker points. The point is that both were shaped for how tweeters are really mounted in cars, not for a flat test panel. The GB15 has a tall, stiff dome in a cup, and the ring around it is a compromise between mounting flush in a flat surface (door top, kick panel) and standing free with nothing around it (sail, pillar). The GB10CF has a "lens" in front that lengthens the path from the dome's center, to avoid the half-wavelength dip described in 2.5 #1. Its stiff dome has a peak at 16 kHz, and the design steers that peak off to the

side, where the longer path inside a car softens it. Recommended setting: high-pass at 2 kHz or higher, LR4 (a steep 24 dB-per-octave filter). This comes from product material (a GB15 review page and Audiofrog's own GB10CF page, as the Sources show), not from a post signed by him.

2.5 #3 One 1" tweeter's dispersion is much like another's.

> RARE Diameter sets the pattern; two 1" domes differ off axis only above about 3 kHz, where stiffness or a center "diffuser" (which can smooth on-axis but hurt off-axis near 15 kHz) matters. Compare data sheets at the same angles; a 15 kHz dip 60° off axis doesn't matter in a car.
>
> **Sources:** DIYMA 2021-11-23

In plain English: "Dispersion" is how widely a speaker spreads its sound, and it depends mostly on the speaker's size compared with the wavelength. Many people believe tweeter brands differ a lot in how widely they spread sound. His point: diameter sets the pattern, so two 1" (25 mm) dome tweeters differ off to the side only above about 3 kHz. Up there, the stiffness of the dome, or a small center "diffuser" placed in front of the dome, can make a difference. A diffuser can smooth the straight-ahead sound but hurt the sound off to the side near 15 kHz. So in practice: when comparing data sheets, compare the graphs at the same angles, and don't worry about a 15 kHz dip measured 60° off axis, because that doesn't matter in a car.

2.5 #4 Tweeters needn't be flush-mounted; keep both listeners within about ±35° of the axis.

> RARE Any driver radiates everywhere at wavelengths long for its size, so a baffle is just a simple dispersion control: "The door isn't a perfectly flat baffle, but that's OK." JBL GTi waveguides were designed for use 35° off axis to suit door mounting.
>
> **Sources:** DIYMA 2011-04-15 · DIYMA 2013-03-22

In plain English: "Flush-mounting" means setting a tweeter so its face is level with the surface around it. The "axis" is the line pointing straight out from the tweeter's center. Many people believe tweeters must be flush-mounted. His point: every speaker spreads its sound in all directions at frequencies whose wavelengths are long compared with its size, so a baffle is just a simple way to control where sound goes. As he says, "The door isn't a perfectly flat baffle, but that's OK." His guideline: keep both the driver and the passenger within about 35° either side of the tweeter's axis. He notes that JBL's GTi tweeter waveguides (shaped surrounds that guide the sound) were designed to be used 35° off axis, to suit mounting in doors.

2.5 #5 Aim only drivers that beam; in a 3-way that's only the tweeter, turned in a little.

> RARE A 2" mid below about 2.7 kHz radiates everywhere, so aiming it changes nothing; aiming a 1" tweeter changes reflections only above about 10 kHz. A tweeter far off axis needs a top-end boost. A friend who preferred his head "between the tweeters" was just closer to them.
>
> **Sources:** Audiofrog blog Jun 2021 · DIYMA 2013-03-10 · DIYMA 2021-10-29

In plain English: "Beaming" is when a speaker playing high notes sends its sound mostly straight ahead in a narrow beam instead of spreading it around. A 3-way system has a tweeter, a midrange and a midbass woofer. His point: aiming only changes something for a speaker that beams at the frequencies it plays. A 2" (50 mm) mid playing below about 2.7 kHz sends sound everywhere, so aiming it changes nothing. Aiming a 1" tweeter changes the reflections only above about 10 kHz. So in a 3-way only the tweeter needs aiming, and only turned in a little. A tweeter that points far away

from you needs a boost in the very top notes to make up for what it loses. And a friend who preferred having his head "between the tweeters"? He explains that the friend was simply closer to them.

2.5 #6 Sail tweeters: crossfire or aim at the listener is a trade.

> There's little difference between 0° and 15°. Crossfiring gives similar levels and a better center for both seats but the image wanders (responses don't match); aiming both at the driver matches responses but adds near-side bias.
>
> **Sources:** DIYMA 2011-11-05

In plain English: A "sail" is the small triangular panel in the front corner of the door window, near the mirror. "Crossfiring" means aiming each tweeter at the opposite seat. The "image" is the illusion of where singers and instruments are; the "center" is a voice that seems to come from straight ahead. His point is that this is a trade-off with no free answer. Turning a tweeter 15° instead of 0° makes little difference anyway. Crossfiring gives similar loudness from both tweeters and a better center for both seats, but the image wanders, because each seat hears one tweeter from a different angle than the other, so their sounds don't match. Aiming both at the driver makes their sound match at the driver's seat, but the nearer tweeter still dominates, pulling the image toward that side ("near-side bias").

2.5 #7 From swivel hardware to "aim only the tweeter, a little".

> **Changed view:** at Harman he co-invented Infinity's swivel flush-mount tweeter (US 5,859,917, filed 1997, about 27° of swivel), and a 2015 JBL/Infinity tip sheet (author unnamed, so not certainly his) said to crossfire pillar tweeters; from 2011 on he prefers sails to pillars and treats aiming only as a treble-balance tool.
>
> **Sources:** Google Patents US5859917A · JBL/Infinity tip sheet · Audiofrog blog Jun 2021

In plain English: This is a changed view. Harman is the company that owned JBL and Infinity, where he worked. While there, he co-invented Infinity's swivel flush-mount tweeter, patented as US 5,859,917 (filed 1997), which let the tweeter turn about 27° so it could be aimed. A 2015 JBL/Infinity tip sheet, which names no author, advised crossfiring tweeters mounted in the pillars (aiming each at the opposite seat). From 2011 on, his own position is different: he prefers sail panels (the small triangles at the front corner of each door window, by the side mirrors) to pillars, and he treats aiming only as a way to adjust how much treble you hear. In short, the swivel hardware and the unsigned tip sheet lean on aiming, while his own teaching since 2011 is to aim only the tweeter, and only a little.

2.5 #8 A tweeter in front of a woofer cone (coaxial-style) warbles.

> RARE The moving cone is the tweeter's baffle, so bass excursion modulates the treble (intermodulation, Doppler-like). Fix by moving the tweeter, high-passing the woofer, or listening off axis. A typical coax has no "phase center" (unlike a KEF Uni-Q); its real benefit is treble in the off-axis sound.
>
> **Sources:** DIYMA 2013-11-29 · DIYMA 2018-04-02

In plain English: A coaxial ("coax") speaker has a tweeter mounted in front of the middle of the woofer cone. "Intermodulation" is distortion in which one sound alters another and creates new, unwanted tones. The Doppler effect is the pitch change you hear as a siren drives past. His point:

in this layout the woofer cone is the tweeter's baffle, so as the cone moves in and out with the bass, it alters the treble, making a warble similar to a Doppler effect. Fixes: move the tweeter somewhere else, high-pass the woofer (so it moves less), or listen from off to the side. He adds that a typical coax does not make all its sound appear to come from one point (a "phase center"), unlike KEF's Uni-Q design. Its real benefit is putting treble into the sound that goes off to the sides.

2.5 #9 Waveguides narrow a tweeter's low-end dispersion to match the woofer; what matters is a smooth directivity index.

> RARE They never widen dispersion (the edge roundover scatters only the highs), and effective ones are big and hard to mount. Narrowing the tweeter's very top to match the woofer makes a power-response hole that "NEVER sounds good"; narrowing its low end is fine if on-axis vs total radiated power stays smooth.
>
> **Sources:** DIYMA 2013-03-22 · DIYMA 2014-04-12

In plain English: A waveguide is a shallow horn shape around a tweeter that controls how widely it spreads sound. A woofer narrows (beams) near the top of its range, while a tweeter spreads very widely at the bottom of its range, so at the handover between them the two can mismatch. "Directivity index" compares the sound straight ahead with the total sound sent out in all directions (the "power response"). His points: waveguides narrow a tweeter's spread at its low end to match the woofer; they never widen it, because the rounded edge only scatters the highest notes. Effective waveguides are big and hard to mount. He separates two cases. Narrowing the tweeter's very top end to match the woofer creates a hole in the power response, which he says "NEVER sounds good". Narrowing its low end is fine, as long as the directivity index stays smooth, changing gradually rather than jumping.

2.5 #10 Use the supplied mounting hardware instead of shop-made rings.

> Tweeters: a coil-spring flush mount (even force all round, unlike leg clips) into a 1" or 1-1/2" hole, a back strap for behind-panel spots, a bendable "tweeter cannon" for pillars. Mids: multi-fit adapter, cast-aluminum ring to machine or bond into panels and upholster, chrome trim ring with room for vinyl, mesh on a damping rubber U-channel. GB60 into wood: plate behind the ring, gasket over the threaded holes, screw speaker and grille through.
>
> **Sources:** DIYMA 2014-10-30 · CMA Networks 2021 @23:03 · DIYMA 2021-12-15

In plain English: His point is to use the mounting parts that come with the speakers rather than having the shop make its own rings. For tweeters, that means: a flush mount held by a coil spring, which presses evenly all the way round (unlike small spring "leg" clips), fitting a 1" (25 mm) or 1-1/2" (38 mm) hole; a strap for mounting behind a panel; and a bendable "tweeter cannon" for pillars. For mids: a multi-fit adapter; a cast-aluminum ring you can machine or glue into a panel and then upholster; a chrome trim ring with room for vinyl; and a mesh grille that sits in a rubber U-channel that damps vibration. For the GB60 (Audiofrog's 6" woofer) going into wood: put a plate behind the ring, put the gasket over the threaded holes, and screw the speaker and grille through.

2.5 #11 Direct-fit OE tweeters: check the critical dimensions first.

> 3D-printed housings put a GB10 or GS10 behind factory grilles (BMW, Audi, Porsche, VW, GM/Bose, Toyota, Tesla and more); GB10OE high-pass 2.5 kHz and GS10OE 3.5 kHz, both 24 dB/oct. Removal tips include reusing BMW's foam cover and unhooking GM Bose tweeters without breaking grille

> tabs. (COMPANY.)
>
> **Sources:** Direct Fit Tweeters and Adapters · GB10OE-0006 sheet

In plain English: "OE" means original equipment: the parts the car came with. "Direct-fit" means made to drop into the factory location. This comes from Audiofrog's own company material. Its 3D-printed housings let a GB10 or GS10 tweeter sit behind the factory grilles in many cars (BMW, Audi, Porsche, VW, GM/Bose, Toyota, Tesla and more). So in practice: check the listed critical dimensions for your car first. The recommended filters are a high-pass at 2.5 kHz for the GB10OE and 3.5 kHz for the GS10OE, both 24 dB per octave (a steep slope). The removal tips include reusing BMW's foam cover, and unhooking GM Bose tweeters without breaking the grille's tabs.

2.6 Choosing drivers and system topology

What this part is about: Choosing speakers and arranging them into a system. A crossover splits the music into frequency bands and sends each band to the speaker suited to it: a "2-way" uses a woofer and a tweeter, a "3-way" adds a midrange between them. It covers where to split, active versus passive crossovers, and how to handle factory systems and processors.

2.6 #1 The key upgrade is a small midrange high in the door, making a 3-way.

> RARE A 2.5" "in just the right spot (high in the door)" may beat a 4" in the wrong one, and with a small mid neither midbass nor mid needs aiming; even cheap OEM 3" mids used only from 900 Hz to 4 kHz bridged a 6" and a tweeter well.
>
> **Sources:** DIYMA 2014-03-04 · DIYMA 2014-04-08

In plain English: His point: the most important upgrade is adding a small midrange high in the door, which turns a 2-way into a 3-way. A 2.5" (63 mm) mid "in just the right spot (high in the door)" may beat a 4" (100 mm) mid in the wrong spot. With a small mid, neither the midbass woofer nor the mid needs aiming, because, as 2.5 #5 explains, neither beams (narrows its sound) in the range it plays. Even cheap factory (OEM) 3" (75 mm) mids, used only from 900 Hz to 4 kHz, bridged the gap between a 6" (150 mm) woofer and a tweeter well. Many people believe a bigger midrange is always better; his point is that the right spot can matter more than size.

2.6 #2 The 6" + 1" door 2-way has a built-in off-axis hole that no EQ fixes.

> RARE A 6" is badly attenuated off axis around 3.5 kHz while most 1" tweeters can't play low enough to meet it; in a car you hear power response (all directions), so the hole is audible even when on-axis is flat, and EQing it makes the direct sound bright. Aiming only moves the hole between direct and reflected sound. Studio monitors get away with 2-ways because the mixer hears mostly direct sound: "The difference is the listening space, not the speakers."
>
> **Sources:** DIYMA 2013-03-10 · DIYMA 2021-11-30 · DIYMA 2022-01-05

In plain English: "Off axis" means to the side of where a speaker points. "Power response" is the total sound a speaker sends out in all directions. EQ (equalization) means boosting or cutting particular frequencies. His point: a 6" (150 mm) woofer sends much less sound to the sides around 3.5 kHz, while most 1" tweeters can't play low enough to take over below that. So off to the sides there is a gap, a "hole", around the handover. In a car you hear the sound from all directions, because so much of it bounces off glass and panels, so the hole is audible even when the straight-

ahead measurement looks flat. Boost it with EQ and the direct sound turns bright. Aiming only moves the hole between the direct and the reflected sound. Studio monitors (the speakers used to mix music) get away with being 2-ways because the person mixing hears mostly direct sound: "The difference is the listening space, not the speakers."

2.6 #3 Fixes for a 2-way's hole: add a small mid, or use a big tweeter.

> "Big tweeter for a 2-way, small tweeter for a 3-way": a tweeter with Fs below about 1 kHz crossed at 1.5–2 kHz (GB15 1.5" + GB60), or a waveguide that narrows the tweeter to match; "in a 3-way any tweeter is fine". No 6" "dispersion widening technology" avoids it: "physically impossible".
>
> **Sources:** DIYMA 2013-03-20 · DIYMA 2014-10-30

In plain English: This follows from 2.6 #2. He gives two fixes for the hole in a 6" plus 1" door 2-way: add a small mid (make it a 3-way), or use a big tweeter. His rule: "Big tweeter for a 2-way, small tweeter for a 3-way". A tweeter whose resonance (Fs) is below about 1 kHz can be crossed at 1.5–2 kHz, low enough to take over before the woofer narrows; his example is Audiofrog's GB15, a 1.5" tweeter, with the GB60, a 6" woofer. Another option is a waveguide that narrows the tweeter's spread to match the woofer. "In a 3-way any tweeter is fine". He says no 6" with "dispersion widening technology" avoids the problem, because that is "physically impossible".

2.6 #4 A crossover's first job is keeping the smaller driver safe: cross every driver except the sub above its resonance.

> RARE Each octave lower costs about 2× power and 4× excursion, and at and below resonance the spider ("parking brake") is the main distortion source; crossed above it, driver distortion is "a nothingburger". Tweeters go at about 1.5–2× Fs. Don't cross a tweeter impossibly low to stop a 6" beaming: build a 3-way.
>
> **Sources:** OFCAP Ep. 19 @1:41:04 · DIYMA 2022-03-13 · DIYMA 2021-08-21

In plain English: Resonance (Fs) is the frequency where a speaker's cone vibrates most easily. An octave is a halving or doubling of frequency, such as 1 kHz down to 500 Hz. "Excursion" is how far the cone moves in and out. The "spider" is the ribbed fabric ring behind the cone that centers it and pulls it back; he calls it the "parking brake". His point: the first job of a crossover is to keep the smaller speaker safe. Each octave lower costs about 2× the power and 4× the excursion. At and below resonance, the spider becomes the main source of distortion; crossed above resonance, the speaker's own distortion is "a nothingburger", meaning negligible. So in practice: cross every speaker except the subwoofer above its resonance, and tweeters at about 1.5–2 times their Fs. Don't cross a tweeter impossibly low to stop a 6" from beaming; build a 3-way instead.

2.6 #5 Its second job: hand over where the low-passed driver still disperses widely.

> RARE Using drivers only where they beam would need about 15" midbass, 10" mids and 4" tweeters. Home-speaker rules still apply in a car because dispersion doesn't change there: matched directivity at each crossover makes off-axis sound resemble on-axis, which makes EQ work (the mic can't separate direct from reflected). He published a chart of dispersion by driver size for picking low-pass points.
>
> **Sources:** DIYMA 2013-03-14 · DIYMA 2021-09-02 · DIYMA 2020-12-23

In plain English: The "low-passed driver" at a crossover is the speaker playing the band below it, such as the woofer under the tweeter. His point: a crossover's second job is to hand over while that lower speaker still spreads its sound widely. He notes that the other approach, using each speaker only where it beams, would need about 15" (380 mm) midbass, 10" (250 mm) mids and 4" (100 mm) tweeters. The rules for home speakers still apply in a car, because a speaker spreads sound the same way there. If both speakers spread sound alike at each crossover, the sound to the sides resembles the sound straight ahead, and that makes EQ work, because the measuring microphone can't tell direct sound from reflected sound. He published a chart of dispersion (how widely sound spreads) by speaker size to help pick these low-pass points.

2.6 #6 His full door recipe.

> RARE Tweeters in the sails, a small mid at the top of the door, the 6" crossed at 400 Hz–1 kHz, everything but the tweeter in its piston range, only the tweeter aimed (a little). It minimizes windshield reflections and maximizes width; add a center that plays to 200 Hz plus steering for two seats. "I can make almost any car sound similar." His MS-8 development Mini took 2nd at IASCA finals this way.
>
> **Sources:** OFCAP Ep. 38 @35:03 · DIYMA 2014-03-30

In plain English: This is his complete front-speaker layout. A "sail" is the small panel in the front corner of the door window. A speaker's "piston range" is the frequencies where its cone moves as one solid piece and spreads sound widely. A "center" is a speaker in the middle of the dash, and "steering" is processing that sends shared, centered sound to it. His recipe: tweeters in the sails, a small mid at the top of the door, the 6" crossed somewhere from 400 Hz to 1 kHz, every speaker except the tweeter kept in its piston range, and only the tweeter aimed, a little. This minimizes reflections off the windshield and makes the stage as wide as possible. For two good seats, add a center that plays down to 200 Hz, plus steering. "I can make almost any car sound similar." His Mini, the development car for the MS-8 (a JBL processor from his Harman years), took 2nd at the IASCA finals, a sound-quality competition, with this layout.

2.6 #7 His crossover points for an Audiofrog 3-way, and why the mid needn't play low.

> Sub 60 Hz; GB60 80–300 Hz; GB25 300 Hz–3 kHz (200 Hz OK, 250 Hz safer; reaching Xmax in IB at 375 Hz LR4 would take about 2,445 W); tweeter above 3 kHz (amp-crossover version: 3.5 kHz). A 4" mid brings no real advantage and needs more box; a big 4" dash mid crossed at 100 Hz only makes sense with no midbass. Raising the GB60/GB25 point usually hides DSP clipping or a leaky pod.
>
> **Sources:** DIYMA 2022-03-10 · CMA Networks 2022 @57:02 · DIYMA 2017-12-19

In plain English: The GB60 is Audiofrog's 6" woofer and the GB25 its 2.5" mid. "Xmax" is how far a cone can travel before it starts to distort badly; "IB" means no box. His suggested split: subwoofer up to 60 Hz; GB60 80–300 Hz; GB25 300 Hz–3 kHz (200 Hz is OK, 250 Hz is safer); tweeter above 3 kHz, or 3.5 kHz when using an amplifier's built-in crossover. He also worked out that pushing the GB25 to its Xmax with no box, crossed at 375 Hz with an LR4 filter, would take about 2,445 W, far beyond its 100 W rating (see 2.4 #2). A 4" mid brings no real advantage and needs a bigger box. A big 4" dash mid crossed at 100 Hz only makes sense when there is no midbass. People who raise the GB60/GB25 crossover point are usually hiding DSP clipping (distortion from overloading the processor) or a leaky pod.

2.6 #8 Use a tweeter: he can't find "a credible argument for not using a tweeter in a car".

> RARE Widebanders were made for desks 18" away: aimed at you they sound "clinical", off axis dull, and only aimed at the windshield do they work. A 3" loses off-axis treble about 1.5 octaves lower than a 1" tweeter; tweeterless systems give tiny images, and "High frequency reflections are pleasant". Pairing "dash widebanders" with a midbass removes the job they were designed for.
>
> **Sources:** DIYMA 2014-04-13 · DIYMA 2015-08-23 · DIYMA 2021-12-02

In plain English: A "widebander" (full-range speaker) is a small speaker, often 2–3", meant to play almost everything with no tweeter. Some enthusiasts prefer a widebander in the dash with no tweeter. His point: he can't find "a credible argument for not using a tweeter in a car". Widebanders were made for desks about 18" (45 cm) from the listener. Aimed at you in a car they sound "clinical", off to the side they sound dull, and they only work when aimed at the windshield. A 3" speaker loses its treble to the sides about 1.5 octaves lower than a 1" tweeter does (1.5 octaves is almost a factor of three in frequency). Systems with no tweeter give tiny images, while "High frequency reflections are pleasant". And pairing "dash widebanders" with a midbass woofer takes away the very job they were designed for: playing the full range alone.

2.6 #9 Put cone breakup and dust-cap peaks out of band.

> A breakup peak comes from a small part of the cone, so it doesn't fall off axis; the dust cap decouples high up with wide dispersion. Diameter sets where off-axis roll-off begins; cone profile matters only via stiffness. Tweeters reach "40 kHz" mainly to push breakup out of the audible band.
>
> **Sources:** DIYMA 2012-09-06

In plain English: "Cone breakup" happens at high frequencies, when a cone stops moving as one piece and parts of it flex, causing peaks. The "dust cap" is the small dome in the middle of a woofer's cone. "Out of band" means outside the range the crossover lets that speaker play. His points: a breakup peak comes from a small part of the cone, so it spreads widely and doesn't fade to the sides. Up high, the dust cap moves separately from the cone and also spreads its sound widely. A cone's diameter sets where its sound starts fading to the sides; the cone's shape matters only through how stiff it is. Tweeters are built to reach "40 kHz" mainly to push their breakup above what we can hear. So in practice: set crossovers so these peaks fall outside each speaker's band.

2.6 #10 Cross midbass-to-mid actively; a passive belongs only between mid and tweeter.

> RARE Audiofrog made no midbass/mid passive ("You should really run an active crossover there"); its hybrid 3-ways pair a mid/tweeter passive with an active midbass split. When the tweeter sits close to the mid, a passive set is where to save money: a 1.5 in offset is half a wavelength at 4.5 kHz, so if the RTA shows a deep narrow dip there, reverse the tweeter.
>
> **Sources:** DIYMA 2014-09-26 · DIYMA 2021-09-26

In plain English: An "active" crossover splits the signal before the amplifiers, usually in a processor (DSP), so each speaker gets its own amp channel. A "passive" crossover is a box of coils and capacitors between one amp channel and the speakers. An RTA (real-time analyzer) uses a microphone to show the level at each frequency. His point: Audiofrog made no passive crossover for the midbass-to-mid split ("You should really run an active crossover there"). Its hybrid 3-ways use a passive between mid and tweeter and an active split between midbass and mid. When the

tweeter sits close to the mid, a passive mid/tweeter set is where to save money. A 1.5" (38 mm) offset between the two is half a wavelength at 4.5 kHz, so if the RTA shows a deep, narrow dip there, reverse the tweeter (swap its + and − wires).

2.6 #11 Car speakers are "NEVER mounted on a flat baffle", so use steep slopes (LR4), not home-style 6 dB/oct overlaps.

> RARE A 6 dB/oct filter's only advantage is phase and it barely protects; cars use more power and compromised placement, so keep the overlap band small. And the "crossover point" is where the measured acoustic curves cross in the car, not the DSP dial.
>
> **Sources:** DIYMA 2014-10-30 · DIYMA 2021-09-27 · DIYMA 2014-08-06

In plain English: A filter's "slope" is how quickly it cuts sound outside its band. 6 dB per octave is gentle and common in home speakers; LR4 is steep (24 dB per octave). The "overlap band" is the range where both speakers play at once, and "phase" describes how their sound waves line up in time. Many people use home-style gentle 6 dB/oct overlaps. His point: car speakers are "NEVER mounted on a flat baffle". A 6 dB/oct filter's only advantage is its phase behavior, and it barely protects the speaker. Cars use more power and compromised placement, so keep the overlap small with steep slopes. Also, the real "crossover point" is where the two speakers' measured curves cross inside the car, not the number set on the DSP.

2.6 #12 Never run a passive network with a section unloaded or split.

> RARE Using only a passive's tweeter side is safe only if the woofer section is a single series inductor; a shunt part can short the amplifier at the network's resonance, and an open tweeter on a 2nd-order network does the same. Splitting a 2-way passive (tweeter side used, mid on its own channel) makes the system cut out at volume. Ohm the tweeter (an Audiofrog one reads 2–3 Ω).
>
> **Sources:** DIYMA 2015-08-27 · DIYMA 2018-01-13 · OFCAP Ep. 19 @2:00:03

In plain English: A passive crossover ("network") is built from coils (inductors) and capacitors. A "series" part sits in line with a speaker; a "shunt" part is wired across the speaker's terminals. "Unloaded" means no speaker connected; "2nd order" means a 12 dB-per-octave filter that uses both a coil and a capacitor. His point: using only the tweeter half of a passive is safe only if the woofer half is a single series coil. If it has a shunt part, at the network's resonance it can short the amplifier. A disconnected (open) tweeter on a 2nd-order network does the same. Splitting a 2-way passive (using its tweeter side while the mid runs on its own channel) makes the system cut out at high volume. So in practice: check the tweeter's resistance with a meter ("ohm" it); an Audiofrog tweeter reads 2–3 Ω (ohms).

2.6 #13 In a series passive crossover the parts swap roles.

> The filter sits in parallel with the driver, so the tweeter's high-pass is a shunt inductor where a parallel network would use a series capacitor.
>
> **Sources:** DIYMA 2022-01-13

In plain English: Most passive crossovers are "parallel": each speaker's filter is wired separately to the amplifier. A "series" crossover chains the speakers instead. A capacitor passes highs and blocks lows; a coil (inductor) passes lows and blocks highs. His point: in a series network, the filter sits in

parallel with (across) the speaker, so the parts swap roles. The tweeter's high-pass becomes a coil wired across it, which drains the low notes away from it. In a parallel network, the same job is done by a capacitor in line with the tweeter.

2.6 #14 Don't band-pass a door woofer as the "mid" of a 3-way from 75 Hz.

> RARE The overlap with the sub kills punchy bass; run the front as a 2-way whose high-pass matches the sub's low-pass. His MS-8 settings: subsonic 20 Hz 12 dB/oct, sub low-pass 80 Hz 24 dB/oct, woofer/tweeter 3.5 kHz 24 dB/oct.
>
> **Sources:** DIYMA 2015-04-21

In plain English: A "band-pass" limits a speaker at both ends, so it only plays a middle band. A "subsonic" filter cuts very deep bass the subwoofer can't use. His point: don't set up a door woofer as the "mid" of a 3-way playing from 75 Hz. Its overlap with the subwoofer kills punchy bass. Instead, run the front as a 2-way (woofer plus tweeter) whose high-pass matches the subwoofer's low-pass. His settings for the MS-8, a JBL processor from his Harman years: subsonic 20 Hz at 12 dB/oct, sub low-pass 80 Hz at 24 dB/oct, and woofer/tweeter split at 3.5 kHz, 24 dB/oct.

2.6 #15 When the factory system is actively crossed, keep its processing and go one-to-one.

> RARE Factory EQ is only about 10–20% speaker correction; Bose/Harman all-pass filters can't be "un-EQ'd", so summing back leaves an unfixable midrange hole. Replace each factory driver with one suited to its channel (GS25 dash, GS60/GS690 door), then add an amp on each factory channel and just retune EQ and levels: no new delays or crossovers.
>
> **Sources:** CMA Networks 2021 @35:00 · Absolute Electronix 2023 @58:34

In plain English: Many factory systems are "actively crossed": the factory amplifier splits the music and sends each speaker its own band, with its own EQ and timing. "All-pass" filters change timing (phase) without changing level. "Summing" means combining the factory channels back into one full-range signal, which many add-on processors do. His point: only about 10–20% of factory EQ is correcting the speakers. Bose and Harman systems use all-pass filters that can't be "un-EQ'd", so summing the channels back together leaves a midrange hole you can't fix. So in practice: keep the factory processing and go one-to-one. Replace each factory speaker with one suited to its channel (for example a GS25 in the dash, a GS60 or GS690 in the door), add an amplifier on each factory channel, and just retune EQ and levels, with no new delays or crossovers.

2.6 #16 For a high-end system, use a separate DSP and amps with analog gain controls.

> RARE A DSP amp's input must squeeze every signal down to its converter's ceiling (about 1–4 V), and after the DAC there is only digital gain, which can't exceed 0 dBFS: after tuning cuts a "1,000 W" amp may deliver about 10 W, with no way back.
>
> **Sources:** Absolute Electronix 2023 @33:35

In plain English: A DSP (digital signal processor) handles EQ, crossovers and timing; a "DSP amp" has one built in. Its input converter turns the music into numbers, and a DAC turns it back into a signal. "0 dBFS" is the highest level a digital system can carry; above it, the sound clips. His point: a DSP amp's input must squeeze every source down to fit its converter's ceiling (about 1–4 V). After

the DAC there is only digital gain, which can't go above 0 dBFS. So once tuning has cut the levels, a "1,000 W" amp may deliver about 10 W, with no way to get the rest back. That is why, for a high-end system, he recommends a separate DSP and amplifiers with analog gain controls.

2.6 #17 There is no perfect speaker, and "point source" is not a shortcut.

> RARE The rules are a century old (enough power handling, flat response across its band, low distortion); change one parameter and "a thousand other things" change, so a speaker is only right for an application. The only practical point source is a headphone; aligning mid and tweeter acoustic centers exactly is "garbage" in a car where heads move.
>
> **Sources:** CMA Networks 2022 @7:02

In plain English: A "point source" is a speaker whose sound all seems to come from a single point. A speaker's "acoustic center" is the spot its sound effectively comes from. Many people treat a point source, or exactly aligning the mid's and tweeter's acoustic centers, as a shortcut to great sound. His point: the rules for a good speaker are a century old: enough power handling, flat response across its band, and low distortion. Change one design parameter and "a thousand other things" change, so a speaker is only right for a particular use; none is perfect. The only practical point source is a headphone. And exact acoustic-center alignment is "garbage" in a car, where listeners' heads move.

2.6 #18 A 3-way he endorsed with big doors and kick mids.

> A user's layout of 9" door woofers at 70–600 Hz, 3.5" kick-panel mids at 600 Hz–2.7 kHz and tweeters above 2.7 kHz got "You're doing it right".
>
> **Sources:** DIYMA 2014-04-02

In plain English: This is a layout a forum user proposed, which he approved with "You're doing it right". It uses 9" (230 mm) woofers in the doors playing 70–600 Hz, 3.5" (90 mm) mids in the kick panels (the side panels by the front footwells) playing 600 Hz–2.7 kHz, and tweeters above 2.7 kHz. It is an example he endorsed, not his own standard recipe.

2.6 #19 A center speaker must play down to at least 200 Hz, and should be able to play loud.

> Use small drivers that reach 200 Hz (a GB25 crossed there works; two 4" drivers let his CES center take a 100 Hz high-pass). His C30 center is a GB60 + GB10, not for bass but because the GB60's 2" coil (vs about 1" on smaller drivers) lets it play loud.
>
> **Sources:** DIYMA 2016-01-18 · DIYMA 2018-03-28 · OFCAP Ep. 38 @1:10:05

In plain English: A center speaker sits in the middle of the dash. The "voice coil" is the coil of wire that drives a speaker's cone; a bigger coil can take more power. His point: a center must play down to at least 200 Hz and should be able to play loud. Use small speakers that reach 200 Hz; a GB25 crossed there works. Two 4" (100 mm) speakers let his CES show car's center use a 100 Hz high-pass. In his Volvo C30, the center is a GB60 (6") plus a GB10 (1" tweeter). That isn't for bass: it is because the GB60's 2" (50 mm) voice coil, versus about 1" on smaller speakers, lets it play loud.

2.7 Audiofrog product-design rationale

What this part is about: Audiofrog is Andy Wehmeyer's speaker and amplifier company. This part gives his reasons for the way its products are designed, built, tested and sold, and what each choice means for the person installing them.

2.7 #1 The brief: easy to put in the right places, no aiming needed, smooth directivity, made to work even for people who skip the manual.

> His list for a great car speaker: low distortion, smooth flat response, uniform off-axis roll-off, easy mounting, designed with its grille. Performance first, then "wrap something beautiful" around it.
>
> **Sources:** DIYMA 2014-03-04 · DIYMA 2017-07-02 · Absolute Electronix 2023 @16:38

In plain English: This is the design goal Audiofrog set itself. The speakers should be easy to mount in the right places in a car. They should not need to be aimed at the listener. Their "directivity" should be smooth: directivity describes how a speaker's sound spreads out to the sides, and smooth means the sound fades gradually, not in jagged steps, as you move off-axis (off to the side). And they should still work well for installers who never read the manual. His list for a great car speaker is low distortion (little added sound that wasn't in the recording), a smooth, flat response (no notes much louder than others), an even fade off to the side, easy mounting, and a grille designed together with the speaker. Performance comes first. Only then do you "wrap something beautiful" around it.

2.7 #2 Car drivers must be designed differently from home drivers.

> RARE Reflections matter more, cars play louder, and the midbass baffle is a corner. Home 6" drivers fake bass with elevated Q ("That higher Q IS distortion"); many car drivers fake it with high inductance. GB uses single-layer flat edge-wound coils, copper pole caps, stiff paper cones and neodymium, which is why a GB60 sounds thin on a display board but full in a car.
>
> **Sources:** DIYMA 2015-12-13 · DIYMA 2016-02-18

In plain English: A "driver" is a single speaker unit: one cone or one dome. Many people believe a good home-audio driver will be just as good in a car. He disagrees. In a car, much of what you hear has bounced off glass and panels, so how a driver sounds off to the side matters more. Cars are also played louder. And the midbass driver usually sits low in a door or kick panel, which acts like a corner and boosts bass (he puts the boost at about 6 dB). Home 6-inch drivers often get extra bass from an elevated "Q", a resonance that makes the bass bump up, and he says "That higher Q IS distortion." Many car drivers fake bass a different way: high inductance (an electrical property of the coil) turns the midrange down so the bass seems louder. Audiofrog's GB drivers use single-layer, flat, edge-wound coils, copper caps on the pole piece, stiff paper cones and neodymium magnets. That is why a GB60 sounds thin on a store display board but full once it is in a car.

2.7 #3 High-end speakers "are not wedding cakes", and trading down should cost output, not accuracy.

> RARE GB took its cue from JBL pro arrays that stay clean at concert levels: loud enough (about 450 W) that owners needn't baby them. Low distortion at high level is what costs money, so GS keeps accurate, smooth sound and gives up output: a smaller budget means "not going to play as loud".
>
> **Sources:** CMA Networks 2023 @14:01 · OFCAP Ep. 19 @1:36:02

In plain English: Many people believe high-end speakers are delicate and shouldn't be played loud. He rejects that: high-end speakers "are not wedding cakes," meaning they shouldn't need careful handling like something fragile. Audiofrog's top GB line took its cue from JBL professional speaker arrays, which stay clean at concert volume, and was made to play loud enough (about 450 W) that owners don't have to baby it. Keeping distortion low at high volume is what costs money. So for the cheaper GS line, Audiofrog kept the accurate, smooth sound and gave up loudness instead. A smaller budget means it is "not going to play as loud", not that it sounds less correct.

2.7 #4 A "work with the factory car" strategy runs through the line.

> Shallow drivers for factory depths, low coax tweeters for factory grilles, gaskets and multi-fit adapters, off-axis voicing for factory spots, inline crossovers that avoid door grommets, full-range drop-ins that keep factory processing (GS25) and 3D-printed OE adapters.
>
> **Sources:** CMA Networks 2023 · BusinessWorld 2024

In plain English: "Factory" or "OE" (original equipment) means the parts the car came with. Many Audiofrog products are designed to fit the car's original speaker spots with little cutting. His examples: drivers shallow enough for factory mounting depths; coaxial tweeters (a coaxial speaker has a tweeter mounted in front of the woofer) that sit low enough to fit behind factory grilles; gaskets and adapters that fit many cars; a tonal balance ("voicing") suited to factory spots that don't point at the listener; inline crossovers that avoid door grommets (a grommet is the rubber boot that carries wires between the door and the car body); full-range drop-in speakers such as the GS25 that let you keep the car's own sound processing; and 3D-printed adapters for factory locations.

2.7 #5 Coaxials are built for low, off-axis door and deck spots.

> RARE A typical coax crosses its tweeter with one capacitor (6 dB/oct), leaving a hole depending on aim; GS coaxes use a 12 dB/oct tweeter network plus 6 dB/oct woofer low-pass on a PCB in the basket, and a waveguide that lifts the tweeter's middle range, so it crosses lower and handles more power. The G62's top end rises to offset the off-axis loss of door-bottom and package-tray spots.
>
> **Sources:** CMA Networks 2023 @8:06 · DIYMA 2021-09-14 · G62 store listing

In plain English: A coaxial ("coax") speaker has a tweeter mounted in front of the woofer. A crossover is the circuit that sends high notes to the tweeter and lower notes to the woofer. Its slope, such as 6 or 12 dB per octave, says how quickly it cuts sound outside each driver's range. He says a typical coax crosses its tweeter with just one capacitor, a gentle 6 dB/oct cut. That leaves a "hole" (a gap) in the sound whose depth depends on where the speaker is aimed. GS coaxes instead use a 12 dB/oct filter on the tweeter plus a 6 dB/oct filter on the woofer, on a small circuit board in the basket (the speaker's frame). They also have a waveguide, a shaped surround that lifts the tweeter's output in the midrange (the lower end of its range), which smooths the handoff from the woofer. So the tweeter can cross over lower and handle more power. On the G62 the treble rises on purpose. That makes up for the treble lost when it sits low in a door or on the rear shelf (the "package tray"), pointing away from your ears.

2.7 #6 The G60S puts each crossover at its driver to keep wires out of the door jamb.

> Woofer: 6 dB/oct low-pass; tweeter: 12 dB/oct high-pass with protection; pre-terminated one-way plugs; the tweeter strips to the bare element for factory or pillar spots. Wire both crossovers' + inputs together and – together to one amp channel (he conceded the manual should say so).
>
> **Sources:** CMA Networks 2021 @44:02 · DIYMA 2021-06-08 · G60S store listing

In plain English: The G60S is an Audiofrog component set: a separate woofer and tweeter, each with its own crossover. Instead of one crossover box, each driver's crossover sits right at the driver. That keeps wires out of the door jamb (the hinge area between door and body). The woofer gets a 6 dB/oct low-pass (it lets lows through and cuts highs). The tweeter gets a 12 dB/oct high-pass (it lets highs through) with built-in protection. The plugs come pre-attached and fit only one way. The tweeter can be stripped down to the bare element to fit factory or pillar spots. Wiring tip: join both crossovers' + inputs together and both – inputs together, and connect them to one amplifier channel. He conceded the manual should say so.

2.7 #7 Drivers are matched by voltage sensitivity: GB mids within about 1 dB, tweeters about 6 dB hotter on purpose.

> RARE Car amps are voltage sources, so 2.83 V ratings predict relative loudness on one amp (the GB25 has lower impedance to keep up with the GB60). The surplus lets a passive attenuate, or an active system turn the tweeter amp's gain down for less hiss and longer tweeter life; GS tweeters are about 3 dB hotter.
>
> **Sources:** DIYMA 2017-10-01 · DIYMA 2015-11-26 · DIYMA 2021-09-04

In plain English: Sensitivity tells you how loud a speaker plays for a given input. Car amplifiers act as voltage sources: they put out a set voltage, and each speaker draws the current it needs. So the useful rating is loudness at 2.83 volts, the voltage that gives 1 watt into 8 ohms. Ratings at 2.83 V predict which speaker will be louder when both are on the same amp. Audiofrog matched its GB mid drivers to within about 1 dB of each other. The GB25 is given a lower impedance (its electrical resistance to the amp's signal) so it keeps up with the GB60. The GB tweeters are made about 6 dB louder on purpose. That spare loudness lets a passive crossover turn the tweeter down. In an active system it lets you turn the tweeter amp's gain down instead, which means less hiss and a longer tweeter life. GS tweeters are about 3 dB hotter than their mids.

2.7 #8 Each tweeter is voiced for its own series; don't mix casually.

> Poly GS cones have a broad, low-Q top-end bump a simple passive can tame, so the GS10 has a matching lower-Q roll-off; the GB10 plays usefully much lower. A 1.5" GB15 at 1 kHz needs a waveguide about 5" across (fine at home, too big in a car), so use a 3-way.
>
> **Sources:** DIYMA 2016-06-02 · DIYMA 2015-12-21

In plain English: Each Audiofrog tweeter is tuned to blend with the woofers of its own series, so he says don't mix them casually. GS woofers have poly (plastic) cones, which have a broad, gentle bump at the top of their range that a simple passive crossover can tame. The GS10 tweeter has a matching gentle ("lower-Q") roll-off at its low end, so the two blend. The GB10 tweeter plays usefully much lower. He also explains why the 1.5-inch GB15 tweeter can't simply be crossed at 1

kHz (1,000 vibrations per second) in a car. It would need a waveguide about 5 inches (about 127 mm) across, which is fine in a home speaker but too big for a car. So in a car, use a 3-way system: tweeter, midrange and midbass.

2.7 #9 The GB25 is a true midrange: pair it with a tweeter.

> Its top-end peak is meant to be removed by the low-pass; running it tweeterless ("please, God no!") works only by EQing that peak out.
>
> **Sources:** DIYMA 2016-03-03

In plain English: The GB25 is Audiofrog's 2.5-inch (about 64 mm) midrange. Its response has a peak at the top of its range. That peak is meant to be removed by the low-pass filter that hands the highs over to a tweeter. Running it with no tweeter at all (his reaction: "please, God no!") works only if you EQ that peak out. EQ, short for equalization, means turning one band of frequencies up or down. So in practice: pair it with a tweeter.

2.7 #10 Why Audiofrog makes no 19 mm car tweeter.

> RARE A smaller dome means a smaller coil and lower power handling, in a market that gives speakers 3× their rating: warranty costs would be massive. Its dispersion gain over a 1" dome is small and falls where music has little content (he made 19 mm tweeters at his previous company).
>
> **Sources:** DIYMA 2022-02-28

In plain English: Some tweeters use a 19 mm (3/4-inch) dome instead of the usual 1-inch (25 mm) one, because a smaller dome spreads sound a little more widely. He explains why Audiofrog makes no 19 mm car tweeter. A smaller dome means a smaller voice coil, which handles less power. And in the car market, people give speakers about three times their power rating, so warranty costs would be massive. The gain in dispersion (how widely sound spreads) over a 1-inch dome is small, and it falls at frequencies where music has little content. He notes that he made 19 mm tweeters at his previous company.

2.7 #11 Stiff cones push the dust-cap peak above the passband for the crossover to remove.

> RARE No competent designer builds narrow "car EQ" wiggles into a driver. The GB60 uses the lightest coil/former and stiffest cone so a steep low-pass removes its cap peak (a phase plug would cost sensitivity); the poly GS60 with a 1" coil turns it into a lower, broader hump needing fewer passive parts.
>
> **Sources:** DIYMA 2015-09-30 · DIYMA 2017-07-24 · DIYMA 2021-09-14

In plain English: At high enough frequencies, every cone stops moving as one stiff piece. That causes a peak in its output, often linked to the dust cap (the dome at the cone's center). He says no competent designer builds narrow "car EQ" wiggles into a driver's response. The real design choice is where that peak lands. The GB60 uses the lightest coil and former (the tube the coil is wound on) and the stiffest cone. That pushes its peak high, above the range the driver is meant to play, where a steep low-pass filter removes it. A phase plug (a fixed bullet in place of the dust cap) would cost sensitivity. The poly-cone GS60, with a 1-inch coil, turns the peak into a lower, broader hump that needs fewer passive crossover parts to tame.

2.7 #12 GB motor and suspension details that matter.

> RARE Single-layer flat CCA coils shed heat from both sides; nearness of coil to steel decides power handling (the neo motor's back cup is the heatsink); flat spiders cut distortion; the flared pole vent reduces port noise. Shorting rings are self-regulating inductance control, not excursion brakes: GB60 has a pole-top copper cap and a base ring (Klippel: distortion "ridiculously low"), mids and tweeters the cap only, subs no ring.
>
> **Sources:** CMA Networks 2021 @17:00 · DIYMA 2015-06-05 · DIYMA 2017-10-28

In plain English: The "motor" is the magnet, steel parts and voice coil that push the cone. The "suspension" is the spider (a flexible, ridged disc behind the cone) and the surround (the flexible ring at the cone's edge). His points: single-layer flat coils of copper-clad aluminum (CCA) shed heat from both sides. How close the coil sits to the steel decides power handling, and in the neodymium motor the back cup is the heatsink. Flat spiders cut distortion. A flared vent through the pole piece reduces port noise (rushing-air sounds). Shorting rings are copper or aluminum rings in the motor. He says they are self-regulating inductance control (they automatically keep the coil's inductance in check), not brakes on how far the cone moves. The GB60 has a copper cap on the pole top plus a ring at the base, and Klippel testing (a speaker measurement system) showed its distortion was "ridiculously low." GB mids and tweeters have the cap only; GB subs have no ring.

2.7 #13 Fancy baskets don't change the sound; a custom basket buys fit.

> RARE Basket "ringing" claims are borderline nonsense; a basket matters only if vents under the spider are too small (mechanical noise). CNC billet is a low-volume cost choice; the real gains are minimal depth, car sizes and room for a flat spider.
>
> **Sources:** DIYMA 2015-09-27

In plain English: The basket is the frame that holds the cone and motor. Many people believe exotic baskets sound better because ordinary ones "ring." He calls basket-ringing claims borderline nonsense. A basket affects the sound only if the vents under the spider are too small, which causes mechanical noise. Baskets machined from solid metal ("CNC billet") are a cost choice for building small quantities, not a sound upgrade. The real gains from a custom basket are fit: minimal depth, sizes that suit car openings, and room for a flat spider.

2.7 #14 Car speakers need car-grade qualification, so Audiofrog uses a Tier-1 automotive factory.

> RARE Tests simulate 50,000 miles: 200-hour heat/humidity cycling (90 °C to −35 °C), thermal shock, UV, 2 g vibration on two axes, drop and package tests, on passives too. Only two factories interviewed would build to that procedure; the Chinese Tier-1 chosen cost more than any other. The inside of a modern door is drier than the cabin.
>
> **Sources:** DIYMA 2015-07-27 · CMA Networks 2021 @46:02 · DIYMA 2020-12-25

In plain English: Parts for cars have to survive heat, cold, damp and shaking for years. He says Audiofrog's speakers are qualified like car-maker parts, with tests that simulate 50,000 miles (about 80,000 km). That means 200 hours of heat and humidity cycling from 90 °C to −35 °C (about 194 °F to −31 °F), thermal shock, UV light, vibration at 2 g (twice the force of gravity) along two axes, and drop and packaging tests. Even the passive crossovers are tested. Only two of the

factories they interviewed would build to that procedure. The Chinese "Tier-1" factory they chose cost more than any other; Tier-1 means a direct supplier to car makers. He also notes that the inside of a modern door is drier than the cabin.

2.7 #15 Speaker quality is measurable, and line-built beats hand-built.

> RARE Quality means meeting spec every time: 100% end-of-line tests (parameters, rub and buzz, polarity, response, distortion). 2015 field data: about 0.28% failures against a 2% limit. 2021 fixes: more glue area under the surround, a positioning jig for slapping GS25 leads, leads soldered both sides. Spec-sheet red flags: missing or 1/3-octave-smoothed curves, sub curves flat to the top.
>
> **Sources:** DIYMA 2015-11-15 · DIYMA 2015-05-30 · DIYMA 2021-06-19

In plain English: He says speaker quality means meeting the specification every time, and that this can be measured. Every Audiofrog speaker is tested at the end of the production line. The tests cover its parameters, rub and buzz (scraping or buzzing noises), polarity (whether the cone moves the right way), frequency response and distortion. Many people assume hand-built speakers are better; he says a tested production line beats hand-building. 2015 field data showed about 0.28% failures, against a 2% limit. Fixes made in 2021: more glue area under the surround, a jig that positions the GS25's lead wires so they don't slap, and leads soldered on both sides. Red flags on a spec sheet: missing response curves, curves smoothed to 1/3 octave (which hides peaks and dips), and subwoofer curves that stay flat right to the top of the graph.

2.7 #16 Passive crossovers are matched to their drivers; don't swap them.

> RARE The GS610C is for GS10 + GS60/GS690, the GS410C for GS10 + GS40/GS25; GS "4 Ω" speakers measure about 2.3–2.8 Ω for factory and 2 Ω-capable amps. The (now discontinued) GB passives' L-pad kept crossover frequency, slope and Q constant over 12 dB. About 95% of GB installs run active vs 60% of GS.
>
> **Sources:** CMA Networks 2021 @30:06 · GB610C page

In plain English: A passive crossover is a set of coils, capacitors and resistors between the amp and the speakers that splits the sound by frequency. Each one is designed for particular drivers. The GS610C is for the GS10 tweeter with GS60 or GS690 woofers. The GS410C is for the GS10 with the GS40 or GS25. GS speakers labelled "4 Ω" (ohms, the unit of electrical resistance) actually measure about 2.3–2.8 Ω. They are designed for factory amps and for amps rated to drive 2 Ω. The GB passives, now discontinued, had an L-pad (a tweeter level control) that kept the crossover frequency, slope and Q the same across a 12 dB range. So turning the tweeter down didn't change the filter. About 95% of GB installs run active (a processor and separate amp channels instead of a passive), versus 60% of GS installs. So in practice: don't swap passives between driver sets.

2.7 #17 Components are sold one piece at a time because dealers threw the passives away.

> Many buyers go active, and he kept seeing boxes of unused passives in install bays; the G series is the all-in-one-box option for DIYers.
>
> **Sources:** CMA Networks 2023 @7:01 · OFCAP Ep. 19 @1:37:04

In plain English: Component speakers come as separate woofers, tweeters and passive crossovers. Audiofrog sells these one piece at a time, and he explains why. Many buyers go "active", meaning each driver gets its own amp channel and a processor does the crossover. He kept seeing boxes of

unused passive crossovers sitting in install bays because dealers threw them away. For do-it-yourselfers who want everything in one box, the G series is the all-in-one option.

2.7 #18 Audiofrog amp crossovers are 12 dB/oct for a mechanical reason.

> RARE A 24 dB/oct pot crossover needs an 8-gang pot whose long shaft twists so sections never track; for 24 dB/oct use a DSP and disable the amp's filters (A150.4) or open them out of band (A600.1).
>
> **Sources:** DIYMA 2022-02-28

In plain English: Many amplifiers have a built-in crossover that you adjust with a knob, called a potentiometer or "pot." A steeper filter needs more sections, and each section needs its own layer, or "gang," on the knob. He says a 24 dB/oct crossover would need an 8-gang pot. Its long shaft twists, so the sections never track (stay matched) as you turn it. That is why Audiofrog amps use 12 dB/oct. For 24 dB/oct, use a DSP (a digital sound processor). Then switch off the amp's own filters (on the A150.4) or set them out of band, away from the frequencies in use (on the A600.1).

2.7 #19 Woofers are designed backwards from the car, so the manual's box is the right box.

> RARE Audiofrog starts from an in-car target curve (averaged transfer functions from about 30 cars) and a box small enough to keep the trunk, then designs the woofer to hit the target in that box. Manufacturers don't publish deliberately "safe" boxes.
>
> **Sources:** Absolute Electronix 2023 @9:49 · DIYMA 2021-09-04

In plain English: A woofer's box changes how it sounds. Audiofrog works backwards from the car. It starts with a target for how bass should measure inside a car, made by averaging the "transfer functions" of about 30 cars. A transfer function is how a car's cabin changes a speaker's sound, especially the bass. Then it picks a box small enough that you keep your trunk space, and designs the woofer to hit the target in that box. So the box in the manual is the right box. Many people believe a manual's box is a cautious "safe" choice. He says manufacturers don't publish deliberately "safe" boxes.

2.8 Driver engineering, ratings and failure diagnosis

What this part is about: This part covers how speaker drivers are engineered and rated, what the numbers on a spec sheet really mean, and how to work out why a speaker is misbehaving or seems to have failed.

2.8 #1 Choose drivers by measurements, not cone material or marketing.

> RARE He judges on- and off-axis response (0–60°), 2nd/3rd-harmonic distortion at small and large signal, and Thiele/Small parameters; buying by material is choosing shoes "based on what the cow ate". A deep cone has no more effective area (Sd) than a flat one; Sd is the cone plus about 1/3–1/2 of the surround. DSP fixes response, not a poor driver's distortion.
>
> **Sources:** DIYMA 2015-09-29 · DIYMA 2021-09-17 · DIYMA 2021-09-15

In plain English: Many people choose speakers by cone material (paper, plastic, metal) or by marketing. He judges them by measurements instead. He looks at frequency response straight on and off to the side (0–60 degrees). He looks at 2nd- and 3rd-harmonic distortion, which means

extra tones at two and three times the note being played, at both low and high volume. And he looks at Thiele/Small parameters, a standard set of numbers describing how a driver behaves at low frequencies. Buying by material, he says, is like choosing shoes "based on what the cow ate." A deep cone pushes no more air than a flat one of the same diameter: its effective area (Sd) is no larger. Sd is the cone plus about 1/3–1/2 of the surround. A DSP (digital sound processor) can fix a driver's frequency response, but it can't fix a poor driver's distortion.

2.8 #2 Compare Xmax only within one method.

> **RARE** Audiofrog rates it as (coil height − gap height)/2 plus 15%: GB60 (20.45 − 5)/2 = 7.75, +15% ≈ "9 mm"; a reviewer's Klippel 10%-distortion method gave about 4.7 mm.
>
> **Sources:** Erin's Audio Corner GB60

In plain English: Xmax is how far a cone can move in one direction while still behaving cleanly. Companies measure it in different ways, so he says compare Xmax figures only when they were measured the same way. Audiofrog's rating takes the voice-coil height minus the magnetic-gap height, halves it, and adds 15%. For the GB60 that is (20.45 − 5) ÷ 2 = 7.75 mm, plus 15%, which gives about "9 mm." A reviewer who used a Klippel method (the distance at which distortion reaches 10%) got about 4.7 mm for the same driver. The difference comes from the method, not the driver.

2.8 #3 How power ratings work: a rating is not a requirement.

> **RARE** "RMS power" should be continuous average power. Audiofrog tests RMS and peak together for 100 hours with shaped noise (average = RMS rating, peaks = peak rating; GB25 at 100 W/300 W down to 200 Hz), so peak isn't invented as 2–3× RMS. A mid or tweeter rating depends on its high-pass; "3× RMS for headroom" is nonsense, and 300 W into a 100 W mid is laughable.
>
> **Sources:** DIYMA 2021-09-20 · CMA Networks 2021 @46:02 · DIYMA 2014-10-30

In plain English: A power rating tells you what a speaker can take, not what it needs. He says "RMS power" should really mean continuous average power. Audiofrog tests the RMS rating and the peak rating together for 100 hours, using shaped noise (a random test signal). Its average level equals the RMS rating, and its peaks reach the peak rating. For example, the GB25 is tested at 100 W average and 300 W peaks, with the signal going down to 200 Hz. So the peak figure is tested, not invented as 2–3 times the RMS figure. A midrange's or tweeter's rating depends on the high-pass filter (which keeps low notes out) used in the test. He calls the advice to buy an amp with 3× the speaker's RMS rating "for headroom" nonsense. Putting 300 W into a 100 W mid, he says, is laughable.

2.8 #4 Phase plug or dust cap: pick by how the driver is used.

> **RARE** A plug's main job is filling the hole left by removing the cap (which would resonate like a bottle); it doesn't raise the beaming frequency. Removing the cap removes its peak but costs area and sensitivity, so behind a low-pass a cap is better.
>
> **Sources:** DIYMA 2015-10-09

In plain English: The dust cap is the dome in the middle of a cone. A phase plug is a fixed, bullet-shaped piece that sits there instead. Many people believe a phase plug helps a driver spread high notes more widely. He says its main job is to fill the hole left when the cap is removed, because an open hole would resonate like blowing across a bottle. It doesn't raise the frequency at which the

driver starts to "beam" (send its sound in a narrow forward beam). Removing the cap gets rid of the cap's peak, but it costs cone area and sensitivity. So for a driver that plays behind a low-pass filter, which cuts off the top of its range, a dust cap is the better choice.

2.8 #5 Where driver distortion comes from, by order.

> RARE Even-order (2nd) comes from asymmetry: a spider pushing unequally, or fringe fields differing above and below the gap. Odd-order comes from symmetric limits, mostly the coil leaving the gap. The spider dominates at and below resonance; motor distortion grows with excursion (inductance change, falling BL). Odd sounds mechanical, even masks detail; 0.2% ≈ −54 dB.
>
> **Sources:** DIYMA 2022-03-16 · DIYMA 2021-12-14

In plain English: Here distortion means extra tones a speaker adds that weren't in the signal. Even-order (2nd) distortion adds a tone at twice the note's frequency; odd-order (3rd and so on) adds tones at three times and other odd multiples. He traces where each comes from. Even-order comes from asymmetry, when something acts differently pushing out than pulling in. Examples are a spider that pushes unequally, or a magnetic field that differs above and below the gap (the narrow slot in the magnet where the coil sits). Odd-order comes from limits that are the same both ways, mostly the coil moving out of the gap. The spider dominates at and below the driver's resonance. Motor distortion grows as the cone moves further, because the coil's inductance changes and BL (the motor's strength) falls. To the ear, odd-order sounds mechanical and even-order masks detail. For scale, 0.2% distortion is about −54 dB: the added tones are 54 dB quieter than the music.

2.8 #6 Break-in is the spider's resin cracking, so it matters for subs, not for correctly crossed mids and tweeters.

> RARE The spider works mainly at and below resonance; heavy resin doping to hit Vas and Q targets is a shortcut that makes parameters drift. Big shifts mean a problem; tweeters have no spider. [Changed view (nuance): 2015 "no break-in needed" → 2017 "real, but it is the spider" → 2021 "real only for subs".]
>
> **Sources:** DIYMA 2015-05-30 · DIYMA 2017-07-28 · DIYMA 2021-11-01

In plain English: Many people believe new speakers must be "broken in" by playing them for hours before they sound right. His view is that what changes is the spider's resin (its stiffening coating) cracking. The spider works mainly at and below the driver's resonance (its natural low frequency). So break-in matters for subwoofers, but not for mids and tweeters crossed over correctly, which never play that low. Some makers dope spiders heavily with resin to hit target Vas and Q numbers (two of the standard low-frequency figures). He calls that a shortcut that makes the numbers drift. Big shifts mean a problem, and tweeters have no spider at all. His view shifted in nuance over time. In 2015 he said no break-in was needed. In 2017 he said it is real, but it is the spider. In 2021 he said it is real only for subs.

2.8 #7 Dual voice coils: series or parallel changes only the load; CCA is fine in coils.

> RARE Motor force, Q, Vas and Fs stay put because $(BL)^2/Re$ scales together; the real cost of dual coils is a wider gap (a DVC can't be single-layer). Spread tinsel leads around, don't bunch them. Coil designers want resistance, so copper-clad aluminum suits coils, unlike power wire.
>
> **Sources:** DIYMA 2022-03-23 · DIYMA 2015-06-16

In plain English: A dual-voice-coil (DVC) woofer has two coils. You can wire them in series (one after the other, giving twice one coil's ohms) or in parallel (side by side, giving half). He says this changes only the load the amplifier sees. The motor force, Q, Vas and Fs (the driver's main low-frequency numbers) stay the same, because the quantity that sets them, $(BL)^2/Re$, scales together either way. BL is the motor's strength and Re is the coil's resistance. The real cost of dual coils is a wider magnetic gap, because a DVC can't be wound in a single layer. Spread the tinsel leads (the flexible wires to the coil) around; don't bunch them. Copper-clad aluminum (CCA) is fine in coils because coil designers want resistance. Power wire is different: there you want as little resistance as possible.

2.8 #8 A narrow dip near 600 Hz that lines up with an impedance wiggle is an "edge hole", a driver flaw.

> RARE It is the surround-to-cone joint resonating; only design changes fix it, so check for it before buying.
>
> **Sources:** DIYMA 2015-07-23

In plain English: On a frequency-response graph, look for a narrow dip near 600 Hz that lines up with a small wiggle on the impedance graph. He calls this an "edge hole," a flaw in the driver itself. It is caused by the joint where the surround meets the cone resonating. Only changing the driver's design fixes it, so check the published graphs for it before you buy.

2.8 #9 Painting cones: check the solvent; cone "treatments" don't help.

> Make sure the paint won't soften the adhesives; EnABL-type treatments measured worse in everything he saw.
>
> **Sources:** DIYMA 2015-10-27

In plain English: If you paint speaker cones, first make sure the paint's solvent won't soften the glues that hold the speaker together. Cone "treatments" such as EnABL-type patterns, applied to a cone in the hope of better sound, measured worse in every case he saw. So he says they don't help.

2.8 #10 Diagnose pops by their sound.

> RARE A sharp pop is probably the former hitting the backplate: stop, repeated hits destroy it. A dull thud is spider or surround bottoming: harmless, just bad-sounding. Also check the surround against the grille (2.1 #19).
>
> **Sources:** DIYMA 2016-05-07

In plain English: He says you can tell what is wrong from the sound of the pop. A sharp pop is probably the former (the tube the voice coil is wound on) hitting the backplate at the rear of the motor. Stop, because repeated hits destroy the speaker. A dull thud is the spider or surround reaching the end of its travel ("bottoming"). That is harmless, but it sounds bad. Also check that the surround isn't hitting the grille (see point 2.1 #19).

2.8 #11 Most "blown" mids and "crackling" tweeters are DSP clipping: check the signal chain first.

> RARE About 70% of GB25 warranty claims were DSP input clipping, 25% broken terminal screws; returned "crackling" units measure perfect. Tweeter crackle when the sub is turned up is digital clipping. If distortion persists with the volume down, check the source (streams and phone levelers clip). Compare two tweeters' DC resistance with a meter.
>
> **Sources:** DIYMA 2022-03-13 · DIYMA 2021-11-14 · Absolute Electronix 2023 @36:46

In plain English: He says most midranges returned as "blown" and tweeters reported as "crackling" are fine. The real cause is usually clipping in the DSP. Clipping happens when a signal is pushed past the most a device can handle, so its peaks get chopped off and it sounds harsh. About 70% of GB25 warranty claims were DSP input clipping, and 25% were broken terminal screws. Returned "crackling" units measure perfect. If the tweeters crackle when you turn the sub up, that is digital clipping. If the distortion stays even with the volume turned down, check the source: streaming services and phone volume "levelers" clip. Use a meter to compare the two tweeters' DC resistance (the basic resistance a multimeter reads). So in practice: check the signal chain before blaming the speakers.

2.8 #12 Fix the car before tuning it, and hunt the one real fault.

> RARE Untunable cars he sees: tweeters left in the dash behind new on-axis mid pods, a midbass wired out of phase "on advice", a split passive. Messy wiring usually isn't the problem; one thing is (a full-rewire bet ended with a broken terminal found in five minutes). GB passives have no polyfuse, so cut-outs mean amp protection or a loose connection.
>
> **Sources:** OFCAP Ep. 19 @1:56:02 · OFCAP Ep. 19 @1:58:00 · DIYMA 2017-07-12

In plain English: "Tuning" means adjusting the processor's settings. He says fix the installation before you tune. Cars he has seen that couldn't be tuned had tweeters left in the dash behind new mid pods aimed at the listener ("on-axis"), a midbass wired out of phase (backwards) "on advice," or a split passive crossover (the notes don't spell out exactly what was done to it). Messy wiring is usually not the problem; one specific fault is. He once made installers a bet over whether a full rewire would fix a car. It ended when a broken speaker terminal was found in five minutes. GB passive crossovers have no polyfuse (a self-resetting fuse that cuts the sound when overloaded). So if a GB system cuts out, suspect the amp's protection circuit or a loose connection.

2.8 #13 A customer who keeps blowing speakers may be boosting bass at the head unit.

> RARE His case: a disco fan (bass-light 1970s mixes) with no sub-level control boosted bass about 12 dB past clipping, burning tweeters and effectively cancelling the midbass high-pass. Fix: raise the crossover, set it up like pro sound, give up front bass he didn't care about.
>
> **Sources:** CMA Connected Oct 2021 @43:34

In plain English: His case is a disco fan whose 1970s recordings are light on bass. His system had no control to raise the sub level, so he boosted bass at the head unit (the car's radio or source unit) by about 12 dB, past the point of clipping. That burned out tweeters and, in effect, cancelled the high-pass filter that was keeping deep bass out of the midbass speakers. The fix was to raise the crossover point and set the system up the way pro sound systems are set up. That meant giving up "front bass" (bass that seems to come from in front of you), which this customer didn't care about.

2.8 #14 Don't buy used drivers online; Audiofrog speakers can't be reconed.

> A stripped terminal screw may be fixable; "maybe you can have it fixed" means walk away.
>
> **Sources:** DIYMA 2020-07-01

In plain English: He advises against buying used drivers online, and Audiofrog speakers can't be reconed (rebuilt with a replacement cone and coil kit). A stripped terminal screw may be fixable. But if a seller says "maybe you can have it fixed," walk away.

2.9 Tweeter protection

What this part is about: Tweeters are the small speakers that play the highest notes, and they are the easiest to damage. This part covers how he says to protect them: protection capacitors, safe crossover settings, and the real reasons tweeters fail.

2.9 #1 Put a series capacitor on tweeters even in active systems: a "parking brake".

> RARE Turn-on pops, a cratering amp, engine noise or a wrong DSP entry can send bass straight to a tweeter, whose coil is about 1/32" long. Example: 68 µF for a 4 Ω tweeter; a $1.75 non-polar electrolytic is fine, since the cap acts below the passband (a 24 dB/oct 2.5 kHz filter is already −48 dB near 600 Hz).
>
> **Sources:** Audiofrog blog Dec 2020 · DIYMA 2015-10-04 · DIYMA 2018-05-04

In plain English: In an "active" system, each driver has its own amp channel and a processor does the crossover, so many people put nothing between the amp and the tweeter. He recommends a capacitor in series with the tweeter anyway, as a "parking brake." A capacitor in series blocks low frequencies and lets highs through. Several things can send bass straight to a tweeter: turn-on pops, an amp that fails ("cratering"), engine noise, or a wrong number typed into the DSP. A tweeter's voice coil is only about 1/32 inch (under 1 mm) long. His example is 68 µF (microfarads, the unit of capacitor size) for a 4 Ω tweeter. A $1.75 non-polar electrolytic capacitor is fine, because the cap acts only below the tweeter's working range, where the crossover has already turned the signal far down. A 24 dB/oct filter at 2.5 kHz is already 48 dB down near 600 Hz.

2.9 #2 How big a protection cap: the documents differ.

> **Conflict:** Tuning PDF v1 (~2018) puts −3 dB about an octave below the tweeter's Fs; the amp manual (c. 2022, COMPANY) says 50 µF plus a resistor, about an octave below the crossover; Tuning PDF v2 (2023) drops the step and relies on LR4.
>
> **Sources:** Tuning PDF v1 (~2018) · A-Series amplifier manual · Tuning PDF v2 (2023)

In plain English: This point carries a [Conflict] tag: Audiofrog's documents give different rules for sizing the protection cap. Tuning PDF v1 (about 2018) says to choose a cap whose −3 dB point is about one octave (half the frequency) below the tweeter's Fs. The −3 dB point is where the signal has been turned down by 3 dB, and Fs is the tweeter's resonance frequency. The amplifier manual (around 2022, a company document) says 50 µF plus a resistor, about an octave below the crossover. Tuning PDF v2 (2023) drops the step altogether and relies on the LR4 crossover filter. LR4 is a common 24 dB/oct filter type (Linkwitz-Riley, 4th order). The point simply records the three versions without settling which is right.

2.9 #3 Turn-on pops through a protection cap: add a resistor across it.

> RARE Amps built on the IR2092 class-D driver chip have a small, unremovable DC offset at turn-on/off; a 10 Ω resistor in parallel with the cap cures the pop (Audiofrog added one to its protectors).
>
> **Sources:** DIYMA 2022-03-16 · DIYMA 2021-07-06

In plain English: Some amplifiers "pop" when switched on or off, and the pop can come through a tweeter even with a protection cap. He says amps built on the IR2092 class-D driver chip have a small DC offset at turn-on and turn-off that can't be removed. Class-D is an efficient amplifier design, and a DC offset is a small steady voltage. A 10 Ω resistor wired in parallel with the cap (across its two ends) cures the pop. Audiofrog added such a resistor to its own tweeter protectors.

2.9 #4 Remove the protection cap when the tweeter goes back on a passive crossover.

> RARE It sits in series with the network's capacitor (series caps combine as 1/(1/C1 + 1/C2)), changing order, slope and Q: it "wacks out" the crossover.
>
> **Sources:** DIYMA 2021-09-22

In plain English: If you switch from active back to a passive crossover, take the protection cap out. The passive network already has its own capacitor in the tweeter's path. Two capacitors in series combine into a smaller value, given by 1/(1/C1 + 1/C2). That changes the crossover's order, slope and Q (how sharply it bends), so it "wacks out" the crossover.

2.9 #5 A protector (TP-50) guards against faults below the crossover, not too much power above it.

> About 15 dB of low-frequency attenuation with no effect in band; it can't stop bad gain setting, an oversized amp or clipping's extra treble, and isn't needed with passives. Wire amp → protector → tweeter. (COMPANY.)
>
> **Sources:** TP-50V2 page

In plain English: The TP-50 is Audiofrog's tweeter protector, and this point comes from the company's product page. It cuts low frequencies by about 15 dB, which guards against faults below the crossover point, and has no effect within the tweeter's working range. It can't protect against badly set gains, an oversized amp, or the extra treble that clipping creates. It isn't needed with passive crossovers. Wire it amp → protector → tweeter.

2.9 #6 Playing at or below Fs is dangerous only for tweeters.

> Their only suspension is the surround, so excess excursion destroys them; other drivers just distort.
>
> **Sources:** DIYMA 2015-06-05

In plain English: Fs is a driver's resonance frequency, where it vibrates most easily. He says playing at or below Fs is dangerous only for tweeters. A tweeter's only suspension is its surround, so too much movement destroys it. Other drivers just distort.

2.9 #7 Tweeter high-pass: about 2× Fs, steep, never lower "because the slope is steep".

> GB10: 2.2 kHz or higher at 24 dB/oct, or 2.5 kHz at 12 dB/oct or more; GB tweeters without the passive: 2.5 kHz LR4 or about 3 kHz with shallower slopes; GB10OE 2.5 kHz and GS10OE 3.5 kHz LR4; GB10CF 2 kHz LR4.
>
> **Sources:** OFCAP Ep. 19 @1:44:09 · DIYMA 2017-10-29 · DIYMA 2021-12-15

In plain English: The high-pass filter decides how low a tweeter is asked to play (kHz means thousands of vibrations per second). His rule: about twice the tweeter's Fs, with a steep slope. Never set it lower just because the slope is steep. His settings: GB10 at 2.2 kHz or higher at 24 dB/oct, or 2.5 kHz at 12 dB/oct or steeper. GB tweeters without their passive: 2.5 kHz LR4 (a standard 24 dB/oct filter), or about 3 kHz with shallower slopes. GB10OE at 2.5 kHz and GS10OE at 3.5 kHz, both LR4. GB10CF at 2 kHz LR4.

2.9 #8 Do steeper slopes protect better?

> **Conflict:** his Oct 2021 blog says steeper slopes don't protect better and crossing lower because the slope is steeper is "a BIG mistake"; his DIYMA posts (2020–21) say LR4 gives high-passed drivers the most protection, "greater even than" 48 dB/oct, without derivation. Practical reading: use LR4 at a safe frequency.
>
> **Sources:** Audiofrog blog Oct 2021 · DIYMA 2020-10-03 · DIYMA 2021-09-27

In plain English: This point records a [Conflict] between two things he wrote. In an October 2021 blog post, he said steeper slopes don't protect better, and that crossing lower because the slope is steeper is "a BIG mistake." In forum posts from 2020–21, he said LR4 (a 24 dB/oct filter type) gives high-passed drivers the most protection, "greater even than" a 48 dB/oct filter, without explaining why. A high-passed driver is one with a filter keeping low notes out. The point's own practical reading is: use LR4, set at a safe frequency.

2.9 #9 Tweeters die from excess high-frequency power, not from "too little power".

> RARE A tweeter gets about a watt; with pink-noise-shaped music, a "20 W" setting gives about 0.16 W at 2.5 kHz and 0.02 W at 20 kHz, so a 200 W amp at full output can kill a passive's tweeter without clipping. Clipping toward a square wave doubles average power, all added as treble; class-D sub amps' output filters spare subs, nothing spares tweeters.
>
> **Sources:** DIYMA 2020-10-14 · DIYMA 2021-07-01 · OFCAP Ep. 19 @27:00

In plain English: Many people believe tweeters are killed by "too little power," meaning a small amp that clips. He says tweeters die from too much high-frequency power. A tweeter gets about a watt. Music has far less energy in the treble than in the bass, and he models this with pink noise, a standard test signal that gets weaker as pitch rises. On his numbers, if an amp is set to "20 W," the tweeter's range gets about 0.16 W at 2.5 kHz and 0.02 W at 20 kHz. So a 200 W amp at full output can kill the tweeter of a passive set without clipping at all. When an amp clips, its output flattens toward a square wave. That doubles the average power, and all of the extra is treble. Output filters in class-D sub amps spare subwoofers from this, but nothing spares tweeters.

2.9 #10 Disconnect tweeters and small mids before setting gains with sine waves: "Never, under any circumstances."

> RARE A "100 W" tweeter is rated for its share of a spectrum calibrated at 100 W at 20 Hz, not 100 W

> continuous; a 0 dB or −10 dB sine track with tweeters connected is the classic killer.
>
> **Sources:** A-Series amplifier manual · DIYMA 2020-12-23

In plain English: A sine wave is a pure, steady test tone, and installers often use one to set amp gains (the amp's input level control). He says to disconnect tweeters and small mids first: "Never, under any circumstances" leave them connected. A tweeter labelled "100 W" is rated for its share of a music-like spectrum calibrated to 100 W at 20 Hz. That share is far less than 100 W, and it is not 100 W of continuous tone. Playing a 0 dB or −10 dB sine track (a full-level or near-full-level tone) with tweeters connected is the classic way to kill them.

2.10 Wiring, termination, polarity and noise

What this part is about: This part covers the wiring side of a system. It explains how to join wires to speakers so the joints last, how to size and fuse power wiring, how to check that every speaker is wired the right way round, and how to track down noise such as engine whine and hiss.

2.10 #1 Crimp, don't solder, wire-to-terminal joints, and crimp ferrules with the proper die.

> RARE Solder can wick up and stiffen tinsel leads, and on GS speakers heat melts the plastic and flows the terminal's joint off the internal PCB (not covered by warranty). Solder terminal to board, crimp wire to terminal; spade lugs last decades. "Cinch terminals are not crimpers": don't let a screw crush a heat-shrunk ferrule.
>
> **Sources:** DIYMA 2021-07-21 · Audiofrog blog Dec 2020 · CMA Networks 2021 @42:00

In plain English: Crimping means squeezing a metal connector tightly onto a wire with a special tool. Many people believe solder always makes the best joint. For wire-to-terminal joints, he says crimp instead. Solder can wick (soak) up the strands and stiffen the tinsel leads (the flexible wires that run to the coil). On GS speakers, soldering heat melts the plastic and makes the terminal's joint flow off the circuit board inside, which the warranty doesn't cover. His rule: solder the terminal to the board, and crimp the wire to the terminal. Spade lugs (flat crimp connectors) last decades. Crimp ferrules (metal sleeves over a wire end) with the proper die, which is the shaped jaw of the crimping tool. "Cinch terminals are not crimpers": don't let a terminal's clamping screw crush a heat-shrunk ferrule into place.

2.10 #2 GB terminal recipe: 16 AWG, strip about 1/4 in, don't tin, push the wire in while backing the screw out, then snug.

> RARE Tinning makes the wire solid so the clamp grips less; under-stripping tempts overtightening. Done right you can swing the speaker by its wires. Don't drill a stripped set screw (shavings reach the gap).
>
> **Sources:** DIYMA 2021-12-18 · DIYMA 2018-06-24

In plain English: GB speakers have screw-clamp terminals, and this is his method for them. Use 16 AWG wire. AWG is the wire-size scale, where smaller numbers mean thicker wire; 16 AWG is about 1.3 mm across. Strip about 1/4 inch (about 6 mm) of insulation. Don't tin it, which means coating the bare strands with solder. Push the wire in while backing the screw out, then snug the screw.

Tinning makes the wire solid, so the clamp grips it less, and stripping too little tempts you to overtighten. Done right, you can swing the speaker by its wires. If a set screw strips, don't drill it out, because shavings reach the gap (the slot in the magnet where the coil moves).

2.10 #3 Power wiring: size it from real music power, fuse near the battery, test grounds by voltage drop.

> RARE Music's average power is a small fraction of peak, so gauge "requirements" from fuse ratings overstate it (a GB active system didn't need 0-gauge), and a short ground run can be thinner. Scrape to bare metal; 60 A fuse within 18" of the battery (COMPANY manual). Check grounds with the battery connected by measuring voltage drop; an ohmmeter can't read a live circuit.
>
> **Sources:** DIYMA 2021-12-22 · A-Series amplifier manual · DIYMA 2021-07-27

In plain English: Power wire runs from the battery to the amplifier. He says size it from the power music really uses. Music's average power is a small fraction of its peaks, so wire "requirements" worked out from fuse ratings overstate what you need. A GB active system he discussed didn't need 0-gauge wire, which is very thick cable, and a short ground wire can be thinner. Scrape the ground point to bare metal. The company's amp manual says to fit a 60 A fuse within 18 inches (about 46 cm) of the battery. Check grounds with the battery connected, by measuring voltage drop: the voltage lost across a connection while current flows. An ohmmeter (a resistance meter) can't read a live circuit.

2.10 #4 Big 3 and stiffening caps: crimp, don't solder, and skip the farad cap.

> Solder near the battery invites corrosion, so crimp Big 3 joints with the right tool; a 2-farad capacitor is unnecessary at 1,000 W. He never said whether a Big 3 upgrade itself is needed.
>
> **Sources:** DIYMA 2021-11-26 · DIYMA 2021-08-02

In plain English: The "Big 3" is a common upgrade to the car's three main charging-system cables. He says crimp Big 3 joints with the right tool rather than soldering them, because solder near the battery invites corrosion. A stiffening capacitor is a large capacitor wired near the amp, here 2 farads. He says one is unnecessary at 1,000 W. He never said whether the Big 3 upgrade itself is needed.

2.10 #5 Verify every driver's polarity, but don't trust a pulse checker on midbass.

> Pulses are too short for low-passed drivers and can't flag an all-pass-filtered factory channel. Mute the others and use band-limited tracks, or play a DC-offset track and read the voltage at the leads with a meter. Sub polarity markings are almost never wrong (3 of ~4,000 on reversible terminals).
>
> **Sources:** DIYMA 2015-06-30 · Audiofrog blog Aug 2021 · DIYMA 2018-04-07

In plain English: Polarity means a speaker's + and − are connected the right way round, so its cone moves out when it should. He says to check every driver's polarity, but not to trust a pulse-type polarity checker on midbass. The pulses are too short for drivers behind a low-pass filter. They also can't flag a factory channel that has an all-pass filter, which shifts the timing of the sound without changing its tone. Instead, mute the other speakers and use band-limited test tracks (tracks covering only that driver's range). Or play a DC-offset track and read the voltage at the speaker leads with a meter. Subwoofer polarity markings are almost never wrong: 3 out of about 4,000 on reversible terminals.

2.10 #6 Some "reversed" mids are correct: baffle placement and crossover order can require it.

> RARE 1st-, 2nd- and 3rd-order slopes shift phase 90°, 180° and 270°, so a passive that inverts a mid "isn't a mistake"; check the summed response before "fixing" it (unlike flipping one side to center vocals, which he forbids).
>
> **Sources:** Audiofrog blog Aug 2021

In plain English: Crossover filters shift phase, which is the timing of a wave within its cycle. He says 1st-, 2nd- and 3rd-order slopes (6, 12 and 18 dB/oct) shift phase by 90°, 180° and 270°. So a passive crossover that connects a mid reversed "isn't a mistake," and where the drivers sit on the baffle can require it too. Measure how the drivers add up together (the summed response) before you "fix" it. That is different from flipping one side's polarity to pull the vocals into the center, which he forbids.

2.10 #7 Match signal cable to the amp's input: ohm the RCA shield to power ground first.

> RARE Near 0 Ω means single-ended (use coax or an LOC); hundreds of ohms or more means differential (use twisted pair, or the head unit's balanced speaker outputs). Twisted pair into single-ended, or coax into differential, lets noise through. Heavy-gauge RCAs are useless; the old "power and signal on opposite sides" rule predates module-filled cars.
>
> **Sources:** DIYMA 2021-09-17 · CMA Networks 2022 @49:00 · DIYMA 2021-07-10

In plain English: Amp inputs come in two kinds. "Single-ended" is the classic RCA input, where one signal wire sits inside a shield that is also the ground. "Differential" uses two signal wires, and the amp responds only to the difference between them, which rejects noise. To tell which your amp has, use an ohmmeter between its RCA shield and its power ground. Near 0 Ω means single-ended: use coaxial RCA cable, or an LOC (line-output converter, which turns speaker-level signals into RCA-level ones). Hundreds of ohms or more means differential: use twisted pair, or the head unit's balanced speaker outputs. Twisted pair into a single-ended input, or coax into a differential one, lets noise through. Heavy-gauge RCAs are useless. The old rule of running power and signal wires down opposite sides of the car predates modern cars full of electronic modules.

2.10 #8 Engine noise: regrounding rarely fixes it now; inputs do.

> RARE Ohm the head unit's RCA shields to each other and chassis, then the amp's to its power ground; if all are grounded the system is built to loop, and only an isolator (or removing the single-ended unit) fixes it. Star grounding dates from all-single-ended days. Avoid single-ended-input amps whatever their heatsinks.
>
> **Sources:** DIYMA 2015-04-18 · DIYMA 2021-11-03

In plain English: Engine noise is a whine that rises and falls with engine speed. Many people believe the cure is regrounding, for example "star grounding," where all grounds meet at one point. He says regrounding rarely fixes it now; the inputs do. Use an ohmmeter to check the head unit's RCA shields against each other and against the chassis, then the amp's RCA shields against its power ground. If they are all grounded, the system is built to form a ground loop, a noise path created by grounds at several points. Then only an isolator (a device that breaks the loop) fixes it, or removing the single-ended unit. Star grounding dates from the days when everything was single-ended. He says to avoid amps with single-ended inputs, whatever their heatsinks look like.

2.10 #9 Hiss is not a ground loop (loops whine); find the source before buying fixes.

> RARE Unplug the inputs: hiss that remains and drops with gain is the amp (DSP amps are inherently hissier); hiss rising with head-unit volume is the head unit. Cure tweeter hiss by lowering that amp's gain and boosting in the DSP, not with series resistors.
>
> **Sources:** DIYMA 2015-09-16 · DIYMA 2021-10-14

In plain English: Hiss is a steady "shhh"; a ground loop makes a whine. He says hiss is not a ground loop, so find where it comes from before buying fixes. Unplug the amp's inputs. If the hiss remains and drops when you turn the amp's gain down, it comes from the amp, and amps with built-in DSP are inherently hissier. If the hiss rises as you turn up the head unit's volume, it comes from the head unit. To cure tweeter hiss, lower that amp's gain and boost the level in the DSP, rather than adding series resistors.

2.10 #10 Noise in one door or only at high excursion: check leads touching metal, then move the passive.

> Tinsel or tweeter leads, or a metal frame, touching body metal cause noise; passive crossovers near noisy electronics pick up radiated noise at low volume.
>
> **Sources:** DIYMA 2015-12-27 · DIYMA 2022-01-02 · DIYMA 2021-08-26

In plain English: If you hear noise in only one door, or only when the speaker moves a lot, he says first check for wires touching metal. Tinsel leads or tweeter wires, or a speaker's metal frame, touching the car's body metal can cause noise. A passive crossover mounted near noisy electronics can pick up radiated noise (interference through the air), heard at low volume. Moving the passive away cures that.

2.10 #11 Class-D amps: shield them from the FM tuner, and know the output filter reacts to the load.

> RARE A class-D output filter is designed for one load, so a 2 Ω load rolls off the top and a rising-impedance tweeter can peak near 20 kHz; Audiofrog's A-series moves it to 40 kHz, matched to its tweeters (flat to almost 30 kHz). Its case and one PCB layer are ground shields, and quiet layout keeps transformer fields from the preamp.
>
> **Sources:** CMA Networks 2022 @35:07 · Absolute Electronix 2023 @29:58 · Audiofrog amplifiers page

In plain English: Class-D amplifiers are efficient amps that switch on and off very fast, and they need an output filter to remove that switching. He makes two points. First, they need shielding from the FM radio tuner. Second, the output filter is designed for one load, so a 2 Ω load rolls off the top end. A tweeter, whose impedance rises at high frequencies, can show a peak near 20 kHz. Audiofrog's A-series amps move that filter up to 40 kHz, matched to its tweeters, which play flat to almost 30 kHz. The amp's case and one layer of its circuit board act as ground shields. A quiet circuit layout keeps the transformer's magnetic fields away from the preamp (the low-level input stage).

2.10 #12 Some factory amps need a load: check before condemning one.

> RARE Many OE amps shut down without a speaker load (to protect their output filter); A-series speaker inputs take up to 30 V peak with no built-in load resistor, so some cars need one. Measure

> the OE output with and without a load first.
>
> **Sources:** DIYMA 2015-04-03 · CMA Networks 2022 @16:01

In plain English: Many factory (OE) amplifiers shut down when no speaker is connected, to protect their output filter. Audiofrog's A-series amps can take the factory speaker wires directly, up to 30 V peak. But they have no built-in load resistor, a resistor that stands in for a speaker, so some cars need one added. Before you decide a factory amp is faulty, measure its output with and without a load.

2.10 #13 Feed a mono sub amp from one left and one right DSP channel.

> RARE Then you can tune the whole right side including the sub, then the whole left, which makes bass and midbass much easier to get right.
>
> **Sources:** DIYMA 2021-08-22

In plain English: A mono sub amp has a single channel. He says to feed it from two DSP channels, one left and one right, rather than from one. Then you can tune the whole right side including the sub, then the whole left side. That makes bass and midbass much easier to get right.

2.11 Top rarely-discussed points in this section

1. With any high-pass, a door midbass needs a sealed baffle, not a door box; tiny sealed boxes are "bass prevention" (2.1 #1, #2).
2. Pull out foam cup baffles; foam rings belong on the bracket and only stop rattles (2.1 #6, #7).
3. The door-plus-corner gain is why a car midbass should be flat and low-Q (2.1 #8, #9).
4. Cross door midbass near 80 Hz and the sub near 60 Hz; 40 Hz midbass wears out the spider and makes a car untunable (2.1 #10).
5. Midbass "punch" lives at 1–4 kHz, not in a bigger woofer (2.1 #13).
6. Leave clearance beyond Xmax: the cone travels through the top-plate thickness too (keep grilles about 1.25 in off a woofer) (2.1 #19).
7. A flat +6 dB with the door trim off is the car's door-switch volume cut, not the panel (2.1 #22).
8. Absorbing a 200 Hz reflection takes about 1.5 ft of material; a 250 Hz peak is usually a reflection, so test with a tone before deadening (2.2 #2, #3).
9. Quiet tires were his best midbass upgrade, and rattles pull the bass image backward (2.2 #4, #7).
10. Measure the distance to the nearest surface before building: it sets the reflection null (6" ≈ 500 Hz; a pod a foot off the pillar corner ≈ a 250 Hz hole) (2.3 #1, #3).
11. A small dash mid aimed at a raked windshield is a "poor man's tweeter"; cut the dash's free low-end gain instead of boosting the top (2.3 #8, #9).
12. Choose locations by acoustic reciprocity: speaker at the driver's head, mic at each candidate spot (2.3 #13).
13. A leaky pod "rattles"; an impedance sweep shows the leak, and at 200 Hz LR4 a small mid barely cares how small its box is (2.4 #2–#4).

14. The 6" + 1" door 2-way has an off-axis hole no EQ fixes; studio monitors get away with 2-ways only because of the room (2.6 #2).
15. In actively crossed factory systems, keep the factory processing and go one-to-one rather than "flattening" the signal (2.6 #15).
16. Most "blown" mids and "crackling" tweeters are DSP clipping, not bad drivers (2.8 #11).
17. Tweeters die from excess treble power (clipping, big amps, sine-wave gain setting), never from "too little power" (2.9 #9, #10).
18. Ohm the RCA shield to pick the right cable; regrounding rarely cures modern engine noise (2.10 #7, #8).

Changed views and conflicts in this section

- **8" door woofers** (2.1 #15): in 2016 Audiofrog would make no 8" and offered the GS690 6×9; the shallow GS8ND2 followed in 2020, framed as a woofer crossed at about 300 Hz, not a midrange.
- **Dash-top center** (2.3 #11): a "frequency response nightmare" in 2010 that still raised the stage, and already his preferred center in the MS-8 FAQ (2011 or earlier; 1.10 #5); by 2021 he advised mounting it as close to the windshield–dash junction as possible, calling its 2 kHz dip helpful. A softening, not a full reversal.
- **Windshield reflections** (2.3 #9): the GS25 deliberately uses the windshield as its tweeter, while his custom door 3-way (2.6 #6) minimizes windshield reflections. An apparent conflict only: one is a factory-location strategy, the other a custom build.
- **Tweeter aiming** (2.5 #7): Harman-era swivel tweeter patent (filed 1997) and a JBL/Infinity tip sheet favouring crossfired pillar tweeters, versus his 2011–2021 advice to use sails and aim only the tweeter, a little.
- **Protection-cap size** (2.9 #2): an octave below the tweeter's Fs (Tuning PDF v1), 50 µF plus a resistor about an octave below the crossover (amp manual), then no cap step at all (Tuning PDF v2).
- **Do steeper slopes protect?** (2.9 #8): his 2021 blog says no; his 2020–21 DIYMA posts say LR4 protects most, even more than 48 dB/oct. Unresolved in his own material.
- **Tweeter high-pass numbers** (2.9 #7): 1.5–2× Fs, about 2× Fs, 2.2 kHz, 2.5 kHz and 3–3.5 kHz appear in different documents; all are well above Fs with steep slopes.
- **Break-in** (2.8 #6): 2015 "no break-in needed" → 2017 "real, but it is the spider, below resonance" → 2021 "real only for subs". A refinement rather than a reversal.
- **Door boxes** (2.1 #1): consistent from 2008 to 2016 (no box with a high-pass; if you build one, size it like a sub box and expect only rattle control).
- **Era notes:** the GB passive crossovers are now listed as discontinued (2.7 #16); MS-8 settings (2.6 #14) are Harman-era.

SECTION 3

3. Tuning and measurement

Andy treats tuning as measuring what is wrong and fixing it in a fixed order: polarity, distance-based delays, standard LR4 crossovers, then level and EQ to a target, with listening only at the end. His tools are cheap (a calibrated mic, free REW software, pink noise and a test CD); what he teaches is how to read the measurement, which errors are audible, and which are just reflections you should leave alone. Subsections 3.1–3.7 cover the process, the mindset, measurement, targets, EQ and crossovers; delay, polarity, level-matching and listening checks follow from 3.8.

Jargon used throughout: **LR4** = Linkwitz-Riley 24 dB/octave crossover; **Fs** = a driver's free-air resonance (**Fc** = resonance in its box); **Q** = how narrow an EQ filter or a peak is (higher Q = narrower); **RTA** = real-time analyzer; **REW** = Room EQ Wizard (free measurement software); **TA** = time alignment (delay); **mono / correlated pink noise** = the same noise in both channels.

What this section is about: "Tuning" means adjusting the electronics that feed each speaker (usually a DSP, a digital signal processor) so the system sounds right from your seat. Andy's approach is to measure with a microphone, fix what the measurement shows in a fixed order, and only listen at the end. This part of the guide covers his step-by-step recipe, the thinking behind it, and the measuring tools he uses. The jargon line above is a glossary; each term is also explained where it first matters below.

3.1 His tuning process, step by step

What this part is about: His complete tuning recipe in the order he does it, from checking the equipment to setting the final volume and personal taste. After the recipe comes a short history of how that order changed between 2008 and 2023.

Master procedure (the order of his Tuning PDF v1 ~2018 and v2 ~2023, with his 2021–22 forum refinements; version differences flagged).

3.1 #1 Check the prerequisites before touching anything.

> The DSP needs delay on every output and separate left and right EQ; its input must be a flat, two-channel, in-phase signal, with input sensitivity matched so the A/D converter never clips. V1 also requires a protection capacitor on each tweeter, even when active. Tools: polarity checker (or the UMI-1 mic plus scope software), REW, the Audiofrog test CD. **Era:** capacitor precondition not restated in v2
>
> **Sources:** Tuning PDF v1 (~2018) · Tuning PDF v2 (~2023) · Audiofrog blog Mar 2021 (scopes and sine waves)

In plain English: Before any adjusting, he checks that the equipment can do the job. A DSP is the box that splits the music to each speaker and lets you adjust each one. He needs it to have delay on every output (each speaker can be held back by a tiny amount of time) and separate EQ for left and right (EQ is a set of very precise tone controls). His PDF says that without separate left and right EQ, "this process isn't going to work very well." The signal going into the DSP must be "flat" (no tone shaping already added), plain left and right, and in phase (both channels pushing the same way), like the output of an aftermarket radio. Its input sensitivity must be set so the loudest signal never overloads ("clips") the A/D converter, the part that turns the incoming signal into numbers.

His 2018 PDF also asks for a protection capacitor on each tweeter (the small treble speaker) even when the DSP does the frequency splitting; a capacitor blocks low frequencies, and he treats it as a safety device. His 2023 version doesn't repeat that requirement. Tools: a polarity checker (or his UMI-1 mic with oscilloscope software), REW, and the Audiofrog test CD.

3.1 #2 Set a safe starting gain structure.

> His 2022 order: set only the sub amp, with a 0 dB sine and the DSP's sub and input levels at maximum, until it just clips; then turn every DSP input and output down 6 dB. The older gain guide found the head unit's maximum clean output first and set all DSP channels to −12 dB. Full gain detail is in section 4. **Changed view: −12 dB → −6 dB**
>
> **Sources:** DIYMA 2022-03-03 · DIYMA "Andy's guide to setting gains" (attributed)

In plain English: "Gain" is the input-sensitivity knob on an amplifier: how hard it amplifies the incoming signal. "Gain structure" is how the volume settings of the radio, DSP and amplifiers are set against each other so nothing distorts and hiss stays low. Clipping is what happens when a device is asked for more than it can give: the tops of the waveform get chopped off, which sounds harsh. His 2022 order: play a 0 dB sine wave (a pure test tone at the maximum digital level) with the DSP's sub output and input levels at maximum, and set only the subwoofer amp's gain so it just clips. Then turn every DSP input and output down 6 dB, and put the test tones away. His older gain guide instead started by finding the head unit's (radio's) maximum clean output, then set all DSP channels to −12 dB. So the starting cut changed from 12 dB to 6 dB over time. Section 4 has the full gain method.

3.1 #3 Confirm every driver's polarity with a checker, the markings or the UMI-1 and a scope.

> Never set polarity by listening for the center image.
>
> **Sources:** Tuning PDF v1 (~2018) · Tuning PDF v2 (~2023)

In plain English: Polarity is which way a speaker's cone moves for a given signal. Every speaker and amp has + and − terminals; if one speaker's wires are swapped, it pushes while the others pull, and their sound partly cancels. He confirms each driver ("driver" just means a single speaker unit) with a polarity checker (a small tool that shows whether the cone moves out or in), with the + and − markings, or with his UMI-1 mic and oscilloscope software, which shows the shape of the signal. His rule is firm: check it with a tool, never by listening to whether the center image sounds right. The center image is the illusion that a singer stands in the middle of the dashboard.

3.1 #4 Strap the mic to the headrest at the listening position, then run the cable out and close the door.

> He measures from outside the car (see 3.4 #1).
>
> **Sources:** Tuning PDF v1 (~2018) · UMI-1 video (Nov 2021) @11:02

In plain English: The measuring microphone goes where your head would be in the driver's seat, strapped to the headrest. He then runs the mic cable out of the car and shuts the door, and works the computer from outside. Section 3.4 #1 gives his reasons: his target curve assumes an empty car, and working from outside spares you a long session of pink noise and keeps the laptop from blocking speakers.

3.1 #5 Tape-measure from each grille or dust-cap center to the mic and enter the delays; then leave them alone.

> The sub is a straight line to the mic whichever way it faces; with passive crossovers measure to the midbass (3-way) or midrange (2-way). Check what your DSP wants: time, distance, or each path subtracted from the longest.
>
> **Sources:** Tuning PDF v2 (~2023) · OFCAP Ep. 19 @3:17:02 · DIYMA 2021-08-26

In plain English: In a car you sit much closer to the speakers on your side of the car than to the ones on the far side, so sound from the near side reaches you first and pulls the image toward it. A DSP can hold back (delay) the nearer speakers so that sound from every speaker arrives at your head at the same moment. This is often called time alignment. He sets it with a tape measure: measure from the center of each speaker's grille or dust cap (the dome in the middle of a woofer cone) to the mic, enter those numbers, and then leave them alone. Sound travels about 34 cm (a little over 1 foot) per millisecond, which is why distance converts directly into delay. The subwoofer is measured as a straight line to the mic, whichever way it faces. If the front speakers share a passive crossover (a small box of parts that splits one amp channel between a woofer and a tweeter), measure to the midbass in a 3-way system or the midrange in a 2-way. DSPs differ in what they want typed in: a time, a distance, or each path subtracted from the longest one. Check yours.

3.1 #6 Enter LR4 crossovers on every driver, with the sub low-pass half an octave below the midbass high-pass.

> Typical: sub 60 Hz low-pass, midbass 80 Hz high-pass, tweeter at about 1.5–2× its Fs, nothing but the sub below its resonance (details in 3.7).
>
> **Sources:** Tuning PDF v1 (~2018) · Tuning PDF v2 (~2023)

In plain English: A crossover splits the music by frequency so each speaker gets only the range it handles well: lowest bass to the sub, upper bass to the midbass, treble to the tweeter. A low-pass filter lets lows through; a high-pass lets highs through. LR4 (Linkwitz-Riley, 24 dB per octave) is a standard, fairly steep filter shape whose two halves add back together smoothly. He uses LR4 on every driver. He sets the sub's low-pass half an octave below the midbass high-pass; his typical numbers are sub low-pass at 60 Hz and midbass high-pass at 80 Hz. He crosses the tweeter at about 1.5 to 2 times its Fs (the frequency where it naturally resonates), and no speaker except the sub is asked to play below its own resonance. Section 3.7 gives the details.

3.1 #7 Calibrate the RTA to the target, then don't touch the volume again.

> Play mono pink noise at normal listening level, offset REW so the midrange sits at 0 dB on the imported Audiofrog target. His 2021 video and the gain guide instead slide the target down onto the response's dips so EQ is mostly cutting (3.6 #3). REW settings and offset steps in 3.3 #5–#6.
>
> **Conflict:** midrange-at-0 dB (PDFs) vs target-on-the-dips (video, forum)
>
> **Sources:** Tuning PDF v2 (~2023) · UMI-1 video @16:00

In plain English: An RTA (real-time analyzer) is a live graph showing how loud each frequency band is at the mic. The target curve is the shape he wants that graph to follow; Audiofrog supplies it as a file you load into REW. Play mono pink noise (the same steady hiss in both channels) at your normal listening level, then shift REW's display so the midrange part of the measured curve sits at

0 dB on the target. After that, don't touch the volume, or the measurement and the target no longer line up. The [Conflict] tag notes two ways he has described this step. In the PDFs, the midrange is put at 0 dB. In his 2021 video and the gain guide, the target is slid down until it sits on the dips (low points) of the measured response, so that most EQ changes are cuts rather than boosts. Section 3.3 #5–#6 gives the exact REW settings.

3.1 #8 Left side plus sub, mono pink noise: get close to the target with channel levels first, then EQ.

> Save the measurement. His PDF example: sub up about 3.5 dB, midrange unchanged, one tweeter up about 6 dB, then EQ.
>
> **Sources:** Tuning PDF v1 (~2018) · DIYMA 2022-03-03

In plain English: Now play mono pink noise through only the left speakers plus the sub, and watch the RTA. First adjust each speaker's overall level (its volume) to get close to the target, and only then use EQ for the finer shape. The gain guide attributed to him says doing levels first gets you partway to the target and makes the EQ faster. A dB (decibel) is the unit for level differences. In his PDF example the sub went up about 3.5 dB, the midrange stayed where it was, and one tweeter went up about 6 dB, before any EQ. Save this measurement, because the right side will be matched to it.

3.1 #9 Right side plus sub: match the saved left within "a couple of dB."

> If one side has a dip you can't fix, copy it to the other side.
>
> **Sources:** Tuning PDF v1 (~2018) · Tuning PDF v2 (~2023)

In plain English: Repeat the same thing for the right speakers plus the sub, matching the right side to the saved left measurement within "a couple of dB." If one side has a dip that can't be fixed, he copies that same dip onto the other side. His reasoning comes from 3.2 #2: matching left to right is the priority, because a matched left and right response is part of his definition of correct stereo.

3.1 #10 Both sides, mono pink noise: a check, not an EQ step.

> Dips of 9 dB or more that appear only in the sum mean wrong delays or delay/phase EQ already in the source. Don't EQ the sum, except a peak above 6 dB cut equally on both sides or a global change such as more bass. Gain guide: set sub level here (about +6 dB if only one input channel feeds the sub).
>
> **Sources:** Tuning PDF v2 (~2023) · DIYMA 2020-07-08 · DIYMA gain guide (attributed)

In plain English: Next, play mono pink noise through both sides at once. This step is only a check, not a place to EQ. If the combined curve shows dips of 9 dB or more that weren't there on either side alone, he says the delays are wrong, or the signal coming into the DSP already has delay or phase-altering EQ in it. Don't EQ this combined curve. The only exceptions are a peak more than 6 dB high, which you cut by the same amount on both sides, and a global change such as more bass. His gain guide sets the sub level at this point; if only one input channel feeds the sub, expect to raise it about 6 dB.

3.1 #11 Now listen: band-limited pink noise, then panned tracks.

> Test-CD tracks 2–5 should each sound centered; tracks 6–20 (pink noise, bass guitar, piano) should move evenly across the dash.
>
> **Sources:** Tuning PDF v1 (~2018) · UMI-1 video @27:01

In plain English: Only now does he listen. Band-limited pink noise is hiss covering just one slice of the frequency range. On the Audiofrog test CD, tracks 2–5 are such slices, and each should sound like it comes from the center of the dash. Tracks 6–20 are pink noise, bass guitar and piano that are "panned," meaning moved gradually from one side to the other. They should move evenly across the dash.

3.1 #12 Center the image with level only.

> Turn down the whole channel on the side the image leans to; never use delay. If the center is smeared or off to one side after the objective steps, suspect a delay-entry error first.
>
> **Sources:** Tuning PDF v2 (~2023) · Absolute Electronix livestream (Mar 2023) @43:32

In plain English: If the center image leans to one side, he fixes it with level only. Turn down the whole channel, meaning all the speakers on the side the image leans toward, by the same amount. Never use delay for this. His 2015 forum advice (3.1 evolution #4) explains the risk: a level error pulls the image, and "fixing" it with delay breaks other frequencies. If, after all the measured steps, the center sounds smeared (spread out) or sits off to one side, his first suspect is a mistake when the delays were typed in.

3.1 #13 If the bass isn't in front:

> look for 60–160 Hz dips, then lower the sub slightly; only as a last resort delay all front speakers by exactly the same amount, gradually, with track 25.
>
> **Sources:** Tuning PDF v1 (~2018) · Tuning PDF v2 (~2023)

In plain English: Sometimes the bass seems to come from behind you, where the subwoofer is, instead of from the front with the rest of the music. His order of fixes: first look for dips between 60 and 160 Hz, around where the sub hands over to the front speakers. Then try lowering the sub's level slightly. Only as a last resort, delay all the front speakers by exactly the same amount, a little at a time, while listening to track 25 of the test CD.

3.1 #14 Set final volume and noise without changing the curve.

> Silent track 38: if hiss bothers you, lower all amp input sensitivities by the same amount; for more volume raise all equally until hiss appears. His 2022 order: raise all DSP outputs equally, then inputs, then every amp gain equally, watching the RTA so the curve doesn't move.
>
> **Sources:** Tuning PDF v1 (~2018) · DIYMA 2022-03-03

In plain English: Track 38 of the CD is silence, used to listen for hiss. If the hiss bothers you, lower all the amplifier input sensitivities (gains) by the same amount. If you need more volume, raise them all equally until hiss appears. Moving everything by the same amount keeps the balance you just tuned. His 2022 order for adding volume is: raise all DSP outputs equally first, then the DSP inputs, then every amplifier gain equally, watching the RTA the whole time so the curve doesn't change shape.

3.1 #15 Preference changes last, through the target or the DSP's input EQ, so both sides change identically.

> Preferences change the target, never the process.
> **Sources:** Tuning PDF v2 (~2023) · DIYMA gain guide (attributed)

In plain English: Personal taste, such as wanting more bass, comes last. He makes those changes through the target curve or through the DSP's input EQ, which acts on the signal before it is split to the speakers, so both sides change identically. His principle: your preference changes the target you aim for, never the steps you follow.

How his procedure evolved

3.1 evolution #1 2008–2009: delay first (with low-passes off), then an RTA to find each driver's usable bandwidth in place, then one channel to the target.

> This manual method was written to copy what the MS-8 automated; 2008 tips add that a measurement 20 dB above background noise is loud enough. **Era:** Harman/MS-8
> **Sources:** DIYMA 2009 · DIYMA 2008

In plain English: In 2008–2009 his manual order was: set the delays first (with the low-pass filters switched off), then use an RTA to find the range each speaker can actually play well where it is mounted, then EQ one channel to the target. He wrote it to copy, by hand, what the MS-8 did automatically. The MS-8 was a JBL car processor from his Harman years that tuned itself with a microphone. His 2008 tips add that a measurement is loud enough when it is 20 dB above the background noise. This item belongs to that Harman/MS-8 period.

3.1 evolution #2 2010–2011: EQ each side alone; in 2011 he even put EQ before delay.

> 2010: EQ right+sub with left off, then left+sub, and largely ignore the both-sides RTA above about 150 Hz. 2011: EQ left, match right (attenuating the nearer speaker), then delay by tape measure.
> RARE **Changed view:** EQ-before-delay, later reversed
> **Sources:** DIYMA 2010 · DIYMA 2011

In plain English: In 2010 he EQ'd each side on its own: the right side plus sub with the left side off, then the left plus sub. He said to largely ignore the both-sides RTA above about 150 Hz. In 2011 he even put EQ before delay: EQ the left side, match the right side to it (turning down the nearer speaker), and only then set delays by tape measure. The tags flag this as unusual and as a view he later reversed; his later process sets delays first. Many people believe you should EQ with both sides playing together; his 2010 method of EQ'ing each side alone went against that.

3.1 evolution #3 2013–2014: delay, crossovers, EQ, recheck, "drink beer."

> Delay comes first in multi-amped systems because it changes the crossover sums; flip a polarity only for a big hole at the crossover.
> **Sources:** DIYMA 2013 · DIYMA 2014

In plain English: By 2013–2014 his order was: delay, crossovers, EQ, recheck, then "drink beer" (his joke for: you're done). His reason for doing delay first in multi-amped systems (one amplifier channel per speaker, split by the DSP) is that delay changes how neighboring speakers add up

around their crossover frequency. He would flip a speaker's polarity only if there was a big hole (deep dip) at the crossover.

3.1 evolution #4 2015: two valid orders.

> Standard: delays by distance, crossovers, then EQ and levels. If you insist on delay by ear, do crossovers and exact EQ first and delay last, because a level error pulls the image and "fixing" it with delay breaks other frequencies.
>
> **Sources:** DIYMA 2015 · DIYMA 2015 (shape drivers to LR4)

In plain English: In 2015 he described two acceptable orders. The standard one: set delays by measured distance, then crossovers, then EQ and levels. If someone insists on setting delay by ear, he said to do crossovers and exact EQ first and delay last. The reason: a level error pulls the image to one side, and if you "fix" that with delay, you break other frequencies.

3.1 evolution #5 MS-8 bring-up (2010–2015): tune a 2-way stereo front first, then add pieces.

> Baseline: subsonic 20 Hz 12 dB/oct, sub 80 Hz 24 dB/oct, front 2-way at 3.5 kHz 24 dB/oct, amp crossovers off, Logic 7 off. Era: MS-8
>
> **Sources:** DIYMA MS-8 FAQ · DIYMA 2015

In plain English: This is how he brought up an MS-8 system (2010–2015): get a simple 2-way stereo front working first, then add the other pieces. His starting settings were a subsonic filter at 20 Hz, 12 dB per octave (a filter that removes the very lowest bass), the sub at 80 Hz, 24 dB per octave, and the front 2-way crossed at 3.5 kHz, 24 dB per octave. Crossovers on the amplifiers were switched off so only the MS-8 did the filtering, and Logic 7 (Harman's surround feature that creates extra channels from stereo) was off. This applies to that one product and period.

3.1 evolution #6 2018 rule for which comes first.

> One EQ per channel: delays first. One EQ per driver: EQ each driver to its crossover target, then delay. The 2018 PDF's process is its first two pages; the next ~60 explain why.
>
> **Sources:** DIYMA 2018-05-22 · Tuning PDF v1 (~2018)

In plain English: His 2018 rule depends on the DSP. If it has one EQ per channel (one EQ shared by all the left speakers, for example), set the delays first. If it has one EQ per driver, first EQ each speaker so its response matches its intended crossover shape, then set the delays. He also notes that his 2018 PDF's actual process fits on its first two pages; the next roughly 60 pages explain why it works.

3.1 evolution #7 2021–2023: same steps, refined.

> His REW guide was written for REW 5.19 (a new one was in progress in 2021); 2022 added the sub-clip-first gain order (step 2); v2 (2023) keeps the steps. By 2023 he said everyone now talks about tuning, so he would teach system design instead (3.2 #15).
>
> **Sources:** DIYMA 2021-07-15 · Tuning PDF v2 (~2023) · Absolute Electronix livestream (Mar 2023) @1:00:14

In plain English: From 2021 to 2023 the steps stayed the same, with small refinements. His REW guide was written for REW version 5.19, and a new guide was in progress in 2021. In 2022 he added the "set the sub amp to clip first" gain order (step 2 above). The 2023 PDF keeps the same steps. By

2023 he said that everyone now talks about tuning, so he would rather teach system design (see 3.2 #15).

3.2 How he thinks about tuning

What this part is about: The mindset behind his process: what "correct" means, why he measures before he listens, why he distrusts tricks and endless ear-tuning, and how he judges whether a tuning method is any good.

3.2 #1 Measure and fix everything objectively first; listen once, at the end.

> He never listens until the measurements say it's worth it, and won't listen after each step ("you don't taste raw chicken after salting it"). He does listen at the end; if something is wrong he goes back to the same instruments, not to random tweaks. Hours of touch-up mean a wrong measurement, a misunderstood one, or a useless process. RARE
>
> **Sources:** DIYMA 2017-06-28 · DIYMA 2021-07-09 · UMI-1 video @24:03

In plain English: His rule is to measure and fix everything that can be measured first, and to listen only once, at the end. He doesn't listen after each step, and compares that habit to tasting raw chicken right after salting it: the dish isn't finished, so the taste tells you nothing useful yet. He does listen at the end. If something sounds wrong, he goes back to the same measuring instruments to find the cause, instead of making random tweaks. If someone needs hours of touch-up by ear, he says the measurement was wrong, was misunderstood, or the process itself is useless. Many people believe tuning is mainly done by ear, listening after every change, which is why this point is tagged as unusual.

3.2 #2 "Correct" stereo has just three conditions: left and right match in level, in frequency response, and in arrival time (phase) at the seat.

> Everything else (tools, targets, tricks) is only useful insofar as it gets you there; a windshield rake or dash mat calls for an adjustment, not a new method.
>
> **Sources:** Mobile Electronics 2017 (tag page) · DIYMA 2017-06-28 · DIYMA 2020-10-03

In plain English: For him, "correct" stereo comes down to three things at your seat: the left and right sides match in loudness, in frequency response (how loud each pitch is, from deep bass to high treble), and in arrival time (phase). Background: when two identical speakers play the same sound at equal loudness and timing, your brain hears it coming from midway between them, which is how a stereo "image" works. In a car you sit off to one side, so tuning restores that match. Tools, target curves and tricks only matter if they get you to those three conditions. A windshield's rake (its slope) or a dash mat calls for an adjustment within the method, not a new method.

3.2 #3 You don't "tune" a system; you measure its deficiencies and fix them.

> He repeats an acoustician friend's line; if the last 20% of your process is choosing which errors you prefer, the objective part needs work. Shops should bill for an efficient, predictable troubleshooting process. RARE
>
> **Sources:** 361 VR Cadillac video @3:01 · DIYMA 2018-05-04 · Mobile Electronics Aug 2017 (DSP Tuning Part 3)

In plain English: He repeats a line from an acoustician friend: you don't "tune" a system, you measure what's wrong with it and fix that. If the last 20% of your process is choosing which errors you prefer to keep, the measured, objective part of your process needs more work. For shops, his view is that they should charge for an efficient, predictable troubleshooting process. Many people believe tuning is a personal art of taste; he treats it more like diagnosing and repairing faults.

3.2 #4 There are no tuning "tricks," only a process; pick one you understand and don't splice in other people's steps.

> A barrel of tips from a contest winner isn't a process, and if someone can't say why a trick works he doesn't know that it works (he posted a 29-step parody of ear-tuning). Mixing teachers gives an "amalgam"; never experiment on customer cars. RARE
>
> **Sources:** DIYMA 2018-07-02 · DIYMA 2021-07-09 · OFCAP Ep. 19 @3:19:02

In plain English: He says there are no tuning "tricks," only a process. Pick one process you understand and follow it, rather than splicing in steps from other people. A collection of tips from a contest winner is not a process, and if someone can't explain why a trick works, he says they don't actually know that it works. (He once posted a 29-step parody of ear-tuning to make the point.) Mixing several teachers' methods gives an "amalgam," a mixture that isn't any one coherent method. And never experiment on customer cars.

3.2 #5 When "the process didn't work," find the one thing done wrong.

> Every failure he has seen came from a skipped or inserted step: a flipped speaker, a 1 kHz peak in one channel, distances typed where the DSP wanted delays. "Cars do strange things" is a cop-out; if tape-measured delays don't give the expected phase, something else is changing phase. RARE
>
> **Sources:** OFCAP Ep. 19 @3:17:02 · DIYMA 2021-08-21 · Absolute Electronix livestream @44:23

In plain English: When someone says "the process didn't work," he looks for the single thing that was done wrong. Every failure he has seen came from a step that was skipped or one that was added. His examples: a speaker wired with flipped polarity, a peak at 1 kHz left in only one channel, or distances typed in where the DSP wanted delay times. He calls "cars do strange things" a cop-out. If delays set by tape measure don't give the expected phase (timing) result, something else in the system is changing the phase, and that is what you should look for.

3.2 #6 Ear-only tuning never converges, because each track excites different frequencies.

> You cut what annoys you on one song, then the next, until "there's no more midrange"; if you keep retuning per track, look for an unfixed high-Q peak. He calls setting things by ear "nuts" and ear-tuning aimless fiddling. RARE
>
> **Sources:** DIYMA 2014 · DIYMA 2015 · DIYMA 2021-05-12

In plain English: "Converge" means settling on a final answer. He says tuning by ear alone never does, because every song emphasizes different frequencies. You cut whatever annoys you on one song, then on the next, and the next, until "there's no more midrange." If you keep re-tuning for each track, he says to look for a narrow (high-Q) peak you haven't fixed yet. He calls setting things by ear "nuts" and describes ear-tuning as aimless fiddling. Many people believe you can tune well by ear alone; this point is his answer to that.

3.2 #7 Judge a method by whether it reliably ends at "correct," then by how teachable and fast it is.

> Repeatability against a defined target is the engineering meaning of quality; for a shop booked six weeks out, a 1-hour process beats a 16-hour one. Tuners who insisted on 10 hours finished his way in about 1 h 20 min; the objective steps take him 30–40 minutes.
>
> **Sources:** DIYMA 2011 · OFCAP Ep. 19 @3:19:02 · Absolute Electronix livestream @43:32

In plain English: He judges a tuning method first by whether it reliably ends up at "correct," then by how easy it is to teach and how fast it is. In engineering, he says, quality means repeatably hitting a defined target. For a shop booked six weeks ahead, a 1-hour process is better than a 16-hour one. He reports that tuners who insisted they needed 10 hours finished his way in about 1 hour 20 minutes, and that the measured, objective steps take him 30–40 minutes.

3.2 #8 Professional tuning needs an agreed endpoint, and you stop when you know what's achievable.

> Billing hours with no defined end is a painter who can't name the color; a sane front-plus-sub system takes him an hour or two. Stop at diminishing returns, not when ideas run out; the threat is "imagine how much better it would sound if…". Don't retune for each recording. RARE
>
> **Sources:** DIYMA 2015 · DIYMA 2022-03-03

In plain English: A professional tune needs an end point agreed in advance. Billing hours with no defined end is, in his comparison, like a painter who can't name the color he's painting. A sensible system of front speakers plus a sub takes him an hour or two. Stop when you reach diminishing returns (more work gives very little improvement), not when you simply run out of ideas. The trap is the endless thought "imagine how much better it would sound if…". And don't re-tune for each recording.

3.2 #9 EQ is the main lever; swapping speakers is "applying a nonadjustable EQ."

> About 90% of problems are EQ-fixable (not distortion); cheap speakers with great EQ beat great speakers with poor EQ, and most complaints about a brand's "voicing" vanish with a DSP. Replace speakers only if broken, underpowered or distorting. RARE
>
> **Sources:** DIYMA 2008 · DIYMA 2016 · DIYMA 2022-01-14

In plain English: EQ (precise tone adjustment in the DSP) is his main tool. He describes swapping speakers as "applying a nonadjustable EQ": a different speaker mostly gives you a different tonal balance, which you can't adjust. He says about 90% of problems can be fixed with EQ (the exception being distortion). Cheap speakers with great EQ beat great speakers with poor EQ, and most complaints about a brand's "voicing" (its typical tonal character) disappear once a DSP is used. He would replace speakers only if they are broken, underpowered or distorting. Many people believe new speakers are the main way to better sound, so this is tagged as unusual.

3.2 #10 The speaker does what it does; everything else you measure at the seat is the car (or a filter you added).

> A two-channel RTA proves it: amp output and the speaker at 1 inch look like the source; the driver's seat with the door shut looks very different. He teaches techs to separate "speaker," "filters" and

> "car" so they can predict any location. RARE
>
> **Sources:** DIYMA 2011 · DIYMA 2021-08-06 · DIYMA 2021-08-06 (troubleshooting)

In plain English: His point is that the speaker itself does what it does; everything else you measure at the seat comes from the car (or from filters you added in the DSP). He says a two-channel RTA, which compares two signals at once, proves it. The amplifier's output, and the speaker measured 1 inch (about 25 mm) away, both look like the source signal. At the driver's seat with the door shut, the curve looks very different. He teaches technicians to separate three things, "speaker," "filters" and "car," so they can predict what will happen in any mounting location.

3.2 #11 A car is not a living room or a concert hall, so imported home and pro methods mislead.

> Speakers and "room" are EQ'd together from one measurement set because seats are fixed and no reflection is late. Imports he mocks: hundredth-of-an-inch delays, 1/48-octave EQ, thousands of averages, first-order "phase-coherent" crossovers, no tweeters, aiming to "eliminate" reflections. RARE
>
> **Sources:** DIYMA MS-8 FAQ · DIYMA 2020-07-08

In plain English: He says a car is nothing like a living room or a concert hall, so methods imported from home audio or professional sound mislead. In a car, the speakers and the "room" (the cabin) are EQ'd together from one set of measurements, because the seats are fixed and no reflection arrives late. (Background: in a large room, echoes from distant walls arrive noticeably after the direct sound; in a car, every surface is close, so reflections arrive almost together with it.) The imported habits he mocks include: delays set to a hundredth of an inch, EQ with 1/48-octave precision, thousands of averaged measurements, gentle first-order "phase-coherent" crossovers, systems with no tweeters, and trying to "eliminate" reflections.

3.2 #12 Cars are more alike than different, so expertise is knowing the general case.

> "Every car is different, just try things" isn't expertise; knowing a DSP's menus isn't knowing audio. Cover "the 95%" and outliers stand out, which is how he predicts a car's problems before listening. Simplicity beats sophistication: complexity isn't useful until it can be explained simply. RARE
>
> **Sources:** DIYMA 2021-12-22 · DIYMA 2017-06-28 · Audiofrog blog Jan 2021 (crossovers)

In plain English: He says cars are more alike than different, so real expertise is knowing the general case. "Every car is different, just try things" is not expertise, and knowing a DSP's menus is not the same as knowing audio. If you understand what is true for "the 95%" of cars, the unusual cases stand out, and that is how he predicts a car's problems before he even listens. He prefers simplicity to sophistication: complexity isn't useful until it can be explained simply.

3.2 #13 His process is a one-seat tune; two-seat all-pass tunes work but he won't teach them.

> The default centers the image for the driver. He sometimes does all-pass two-seat tunes, but they take fiddling and listening, so they aren't a repeatable process. On a later podcast he called "forget the passenger" a cop-out (details from 3.8). **Changed view:** one-seat (PDFs) → two-seat where tools allow (2016 onward)
>
> **Sources:** Tuning PDF v1 (~2018) · DIYMA 2021-09-07 · OFCAP Ep. 38 @52:04

In plain English: His written process tunes for one seat, the driver's, so the image is centered for the driver. "Two-seat" tuning tries to make the image work for driver and passenger at once, often with all-pass filters (DSP filters that change the timing of some frequencies without changing their loudness). He says these can work and he sometimes does them, but they need fiddling and listening, so they are not a repeatable process and he won't teach them. The [Changed view] tag records two positions: his PDFs present a one-seat tune, while from 2016 onward, where the DSP's tools allow it, he moved toward tuning for both seats, and on a later podcast he called "forget the passenger" a cop-out. Section 3.8 has the details.

3.2 #14 "Correct" is the beginning of expertise, not the end.

> Learn to reach correct, then learn to back off it for customers who want something else; a contest win only proves judges liked it. Spatial errors bother him more than tonal ones: a missing center image is worse than a peak or dip. RARE
>
> **Sources:** CMA Connected (Oct 2021) @42:16 · DIYMA 2016 · SQOLOGY P2 @46:07

In plain English: Reaching "correct" is where expertise starts, not where it ends. First learn to get a system to correct; then learn how to move away from it on purpose for a customer who wants something different. Winning a contest only proves that the judges liked it. He says errors in where sounds appear (spatial errors) bother him more than errors in tone: a missing center image is worse to him than a peak or a dip in the response.

3.2 #15 Tuning begins with system design; by 2023 he taught mapping the car with a mic and speaker before choosing locations.

> Measure every candidate spot (dash top, A-pillar, kick, door top, trunk) instead of guessing rules like "avoid the windshield corner"; he did this for his son's car and the Volvo C30 demo car. RARE Era: 2021–2025
>
> **Sources:** CMA Connected (Oct 2021) @41:52 · Absolute Electronix livestream (Mar 2023) @1:00:14 · OFCAP Ep. 38 @36:00

In plain English: He says tuning begins with system design: deciding where the speakers go. By 2023 he taught mapping the car first, by placing a speaker in each candidate spot (dash top, A-pillar, kick panel, door top, trunk) and measuring it with a mic before choosing locations. (The A-pillar is the post beside the windshield; the kick panel is the side panel near your feet.) He prefers measuring to following rules of thumb such as "avoid the windshield corner." He did this for his son's car and for the Volvo C30 demo car. This reflects his teaching from about 2021–2025.

3.2 #16 Train your ears on two speakers in phase vs out of phase, not on concerts.

> Many tuners have never sat between two speakers and flipped one's polarity. Practice with any matched $5 thrift-store pair in an equilateral triangle: imaging is just what two identical speakers do for a centered listener. RARE
>
> **Sources:** Absolute Electronix livestream @41:56 · Absolute Electronix livestream @46:15

In plain English: He says to train your ears on two speakers playing in phase versus out of phase, not by going to concerts. Many tuners, he says, have never sat between two speakers and flipped the polarity of one. Practice is cheap: any matched pair of $5 thrift-store speakers, set up so you and the two speakers form an equilateral triangle (all three sides the same length). His point is that

imaging is simply what two identical speakers do for a listener sitting in the center, so hearing that effect directly teaches you what to listen for. Many people believe the best ear training is listening to live concerts; that is the common advice he disagrees with.

3.2 #17 An auto-tune is a spec plus a hit rate.

> Judge any automatic optimizer by how often it hits its target across systems; the MS-8 worked within its 48 supported layouts and broke on edge cases, and no auto-tune beats every manual process every time. **Era:** Harman/MS-8
>
> **Sources:** DIYMA 2016

In plain English: An auto-tune is a processor that measures and tunes the system by itself. He says to judge any such optimizer by its specification plus its hit rate: how often it reaches its target across many different systems. The MS-8 (the JBL self-tuning processor from his Harman years) worked well within the 48 speaker layouts it supported and broke on unusual edge cases. He adds that no auto-tune beats every manual process every time. This point relates to the MS-8 era.

3.3 Measurement tools and setup

What this part is about: The measuring kit and how to set it up: which test signal to use, which microphone, how to set up the free REW software, and the common ways a measurement can mislead you.

3.3 #1 For production tuning use a live RTA on mono pink noise, not sweeps or sine tones.

> An RTA shows the curve move as you EQ; a sweep or a tone excites each narrow comb peak and dip one at a time and misleads in a car ("Pure tones aren't good tools"). He still uses REW sweeps for research and phase. **Changed view:** in 2012 he thought REW sweeps worked "much better"
>
> **Sources:** Audiofrog blog Jan 2021 (single vs multi-mic) · DIYMA 2021-09-03 · DIYMA 2012

In plain English: "Production tuning" means everyday tuning of customer cars in a shop. For that he uses an RTA (a live graph of loudness across frequencies) fed with mono pink noise (steady hiss, the same in both channels). The RTA updates live, so you see the curve move as you change the EQ. A sweep (a tone that glides from low to high pitch) or single test tones hit the narrow peaks and dips one at a time and give a misleading picture in a car. He sums it up as "Pure tones aren't good tools." Background: those narrow peaks and dips come from comb filtering, which happens when the same sound reaches the mic twice with a tiny time gap (directly and off a nearby surface), so some frequencies add up and others cancel. He still uses REW sweeps for research and for measuring phase (timing). The [Changed view] tag notes that in 2012 he thought REW sweeps worked "much better."

3.3 #2 Pink noise, not white.

> Pink noise has equal power per octave, sounds even from 20 Hz to 20 kHz and resembles music's energy spread; the RTA shows it flat by applying the inverse 3 dB/octave slope. White noise sounds bright and isn't used for car tuning.
>
> **Sources:** Audiofrog blog Dec 2020 (pink noise)

In plain English: Pink noise is a hiss with equal power in every octave (an octave is a doubling of frequency, such as 100 to 200 Hz or 1,000 to 2,000 Hz). It sounds even from 20 Hz to 20 kHz and resembles how music's energy is spread. On many analyzers it would show as a line sloping downward, so the RTA tilts the display the opposite way (the "inverse 3 dB/octave slope") to make pink noise read flat. White noise has equal power at every single frequency, which piles up energy in the treble, so it sounds bright. He says white noise isn't used for car tuning.

3.3 #3 Never tune or center the image with 31 sine waves; if you must use your ears, use band-limited pink noise or music.

> Above 1 kHz the comb filtering is so dense that a tone easily lands in a null, and the nulls differ left to right; that's why the 1980s pure-tone centering method fails ("really stupid"). RARE
>
> **Sources:** DIYMA 2021-07-08 · DIYMA 2017-07-12 · DIYMA 2020-07-08

In plain English: An old method was to play 31 pure tones (one for each band of a classic 31-band EQ) and adjust until each sounded centered. He says never tune or center the image this way. If you must use your ears, use band-limited pink noise (hiss covering only a slice of the range) or music. His reason: above 1 kHz the comb filtering (narrow cancellations from reflections) is so dense that a single tone easily lands in a null, a spot where it almost cancels, and the nulls fall at different frequencies on the left and right. That is why the 1980s pure-tone centering method fails; he calls it "really stupid." Many old-school competitors believed tone-by-tone centering was the right way, which is why this point is tagged as unusual.

3.3 #4 Use mono (correlated) noise; uncorrelated noise has no center, so it can't show the summed image.

> With one side playing the noise type doesn't matter; with both, you're measuring their sum. Uncorrelated noise is only for making everything but the center play in an upmixer, driving an L−R rear feed, or "gaming" an IASCA RTA. To measure L−R rears, play one channel (or pan mono noise fully). RARE **Changed view:** in 2014 he suggested stereo noise for 31-band adjustments
>
> **Sources:** DIYMA 2021-10-08 · DIYMA 2020-07-08 · DIYMA 2014

In plain English: Correlated (mono) noise is the identical signal in both channels, so with both sides playing it forms a sound in the center. Uncorrelated noise is a different random hiss in each channel, so it has nothing in common between the sides and no center. He says to use mono noise: with one side playing, the type of noise doesn't matter, but with both sides playing you are measuring their sum, and only correlated noise shows that. Uncorrelated noise has only a few uses: making every speaker except the center play in an upmixer (a processor that creates center and rear channels from stereo), driving rear speakers fed with L−R (the difference between left and right), or "gaming" the RTA test in IASCA competitions. To measure L−R rears, play one channel only, or pan mono noise fully to one side. The [Changed view] tag notes that in 2014 he suggested stereo noise for adjusting a 31-band EQ.

3.3 #5 His REW RTA settings.

> PDFs: 1/3-octave RTA (1/12 to measure a peak's width); FFT 16384, 16 averages, rectangular window, 50% overlap, update interval 2; 5 dB per division; 20 Hz–20 kHz; bars on. In 2021 he said use 32k, window and 64k make no difference, and dB vs dBFS doesn't matter. **Changed view:** FFT 16k → 32k; "dB not

> dBFS" → either
>
> **Sources:** Tuning PDF v2 (~2023) · DIYMA 2021-12-05 · DIYMA 2021-07-15

In plain English: These are his settings for REW's RTA. In his PDFs: 1/3-octave display (switch to 1/12-octave when you need to measure how wide a peak is); FFT length 16384; 16 averages; rectangular window; 50% overlap; update interval 2; 5 dB per division on the graph; range 20 Hz to 20 kHz; bars turned on. Background: FFT length is how big a chunk of sound the analyzer studies at once (bigger means finer detail); averages smooth out the jumpy display; dB per division is the vertical scale of the grid. In 2021 he said to use 32k, that the window type and a 64k length make no difference, and that showing dB or dBFS (decibels relative to digital full scale) doesn't matter. Those are the changes the [Changed view] tag records.

3.3 #6 Display calibration in REW.

> Read the pink-noise level midway between the loudest and quietest bands (his example 75 dB), enter it under Measurement Offsets "All SPL," click "Add Offset to Data," then import "Audiofrog Target Curve for REW.csv" (answer "No") and check the midrange sits at 0 dB. For "whack-a-mole" hunting he once preferred 3 dB per division.
>
> **Sources:** Tuning PDF v1 (~2018) · DIYMA 2008

In plain English: This sets REW's display so your measurement can be compared with the target curve. Play pink noise and read the level midway between the loudest and quietest bands; his example is 75 dB. Type that number into Measurement Offsets, "All SPL," and click "Add Offset to Data," which shifts the measurement so it sits around 0 dB. Then import the file "Audiofrog Target Curve for REW.csv," answer "No" to the prompt that appears, and check that the midrange sits at 0 dB. "Whack-a-mole" tuning is his name for the problem where fixing the sound on one track breaks another. To hunt down the serious peaks and dips behind it, he once (around 2008–2009) preferred a finer graph scale of 3 dB per division instead of the default 10.

3.3 #7 You need a known mic, not an expensive one.

> Software applies the mic's calibration file, so a cheap calibrated capsule equals a precision mic (his first Harman rig was a ~$6,000 DOS "lunchbox"). His 11 mics, from ACO Pacific and Earthworks to UMIK-1, ECM8000 and UMI-1, read within about 1 dB except at the very top; avoid the APEX 220. A "true RTA" is just an RTA. RARE
>
> **Sources:** DIYMA 2021-08-16 · UMI-1 video @9:01 · DIYMA 2016

In plain English: He says you need a microphone whose errors are known, not an expensive one. Each measurement mic can come with a calibration file listing its own small errors, and the software corrects for them, so a cheap calibrated capsule measures as well as a precision mic. For contrast, his first measurement rig at Harman was a roughly $6,000 DOS-computer "lunchbox." He owns 11 mics, from ACO Pacific and Earthworks to the UMIK-1, ECM8000 and UMI-1, and they read within about 1 dB of each other except at the very top of the range. He says to avoid the APEX 220. He adds that a "true RTA" is just an RTA. Many people believe good measurements need an expensive mic or special analyzer, so this is tagged as unusual.

3.3 #8 No 90° calibration file is needed in a car, and pointing direction hardly matters.

> A car is a pressure field at low frequencies and random-incidence (diffuse) at high ones, so an omni pointed up or forward is fine; 90° files are for free-field work. Most such mics use similar 1/4-in electret capsules, and max SPL is set by bias voltage (5 V USB gives a bit over 100 dB). RARE
>
> **Sources:** DIYMA 2020-10-03 · DIYMA 2021-10-26

In plain English: Some calibration files come in a 0° version (mic pointed at the speaker) and a 90° version (mic pointed sideways or up). He says the 90° file isn't needed in a car and the direction the mic points hardly matters. His reason: at low frequencies a car cabin is a "pressure field," where the sound pressure is about the same everywhere in the small space. At high frequencies it is "random-incidence" (diffuse), meaning sound arrives from all directions at once after bouncing around. So an omni mic (one that picks up from all directions) pointed up or forward is fine; 90° files are for "free-field" work, where sound comes from one direction in open space. Most such mics use similar 1/4-inch (about 6 mm) electret capsules, and the loudest sound they can handle (max SPL) depends on their bias voltage: 5 V from USB gives a bit over 100 dB.

3.3 #9 A phone's built-in mic is usable at low and mid frequencies.

> Around 300 Hz it is close enough to omnidirectional; a real measurement mic matters mainly at high frequencies. RARE
>
> **Sources:** DIYMA 2021-11-02

In plain English: He says even a phone's built-in microphone is usable for low and middle frequencies. Around 300 Hz it picks up from all directions closely enough to work. A real measurement mic matters mainly for the high frequencies.

3.3 #10 Your playback device can corrupt the test signal; use an external sound card and the right calibration.

> Many laptops apply speaker protection in the OS that can't be disabled, Bluetooth may tap the signal after it, and outputs can clip near full volume. REW's soundcard calibration covers input and output together, so clear it when the pink noise comes from the CD or a phone. RARE
>
> **Sources:** DIYMA 2021-12-22 · DIYMA 2018-05-17 · Testgear site

In plain English: The test signal can be spoiled before it ever reaches the speakers. Many laptops apply "speaker protection" processing in the operating system that can't be turned off, Bluetooth may take its signal after that processing, and outputs can clip (distort) near full volume. So he recommends an external sound card (a separate audio interface). REW's soundcard calibration corrects the computer's input and output together, so if the pink noise is coming from the CD or a phone instead of the computer, clear that calibration, since the computer's output isn't part of the chain then.

3.3 #11 "Nothing above 10 kHz" is usually the source.

> A 128 kbps rip, Bluetooth SBC (cuts around 12–15 kHz) or MP3 drops the top octave, so don't EQ or judge highs from them. Lossy codecs shouldn't touch bass, so missing bass over Bluetooth means another fault.
>
> **Sources:** DIYMA 2021-12-05 · Testgear site

In plain English: If your measurement shows nothing above 10 kHz, he says the cause is usually the source, not the speakers. A 128 kbps rip, Bluetooth's basic SBC codec (which cuts off around 12–15 kHz) or an MP3 can drop the top octave of treble. So don't EQ or judge the highs using those sources. These "lossy" formats save space by discarding sound, but he says they shouldn't affect bass, so if bass goes missing over Bluetooth, look for a different fault.

3.3 #12 Gating and frequency-dependent windows are useless in a car.

> Reflections arrive too early to be heard as echoes (no Haas effect, no reverberant field, no late reflections), so they act as frequency-response errors you hear. A window short enough to exclude them loses all low-frequency resolution; gated data show nothing useful below about 1 kHz. RARE
>
> **Sources:** DIYMA 2009 · DIYMA 2014 · DIYMA 2017-10-28

In plain English: "Gating" or "windowing" means measuring only the first few thousandths of a second after the sound leaves the speaker, to cut out reflections. It is common in home speaker design. He says it is useless in a car. In a big room, a reflection can arrive late enough to be heard as a separate echo or as room sound; the Haas (precedence) effect and the reverberant field are the names for how the ear handles those later arrivals. In a car, every reflection arrives so soon after the direct sound that none of that happens; instead, reflections simply change the tone, as frequency-response errors you hear. A window short enough to exclude them is too short to capture slow bass waves, so it loses all low-frequency detail; he says gated data show nothing useful below about 1 kHz. Many people believe gated measurements, a standard home-speaker technique, also work in cars, which is why the point is tagged as unusual.

3.3 #13 Scopes and 0 dB sine waves are of little use for setting DSP-system levels.

> Setting mid and tweeter amp levels with a sine and scope is "a colossal waste of time" and can kill tweeters; doing it before the acoustic tune is riskier than setting gains by ear. Use them only for clipping points (gain detail in section 4). RARE
>
> **Sources:** Audiofrog blog Mar 2021 (scopes and sine waves) · Audiofrog blog Dec 2020 (pink noise)

In plain English: Many installers believe the proper way to set amplifier gains is with an oscilloscope (a tool that draws the waveform) and a 0 dB sine wave (a full-level test tone). He says this is of little use in a DSP system. Setting the midrange and tweeter amplifiers this way is "a colossal waste of time" and can kill (destroy) tweeters. Doing it before the acoustic tune is, in his view, riskier than simply setting gains by ear. He says to use scopes and sine waves only to find clipping points; section 4 has the gain details.

3.3 #14 Phase and timing measurements need a two-input interface (loopback) or an acoustic reference.

> Put the loopback on one input and the mic on the other; USB-only mics can't do it. Without loopback, drive the nearest tweeter as REW's acoustic timing reference; Audiofrog's PAA-1 preamp lets a UMI-1 capsule run on a 2-channel interface.
>
> **Sources:** DIYMA 2009 · DIYMA 2017-06-28 · DIYMA 2021-10-26

In plain English: Measuring timing and phase needs a reference for when the signal was sent. One way is "loopback": a two-input sound interface where one input receives the test signal straight from the output by cable, and the other input gets the mic. USB-only mics can't do this, because

they have just one input. Without loopback, REW can use an "acoustic timing reference" instead: play a signal through the nearest tweeter first and time everything from it. Audiofrog's PAA-1 preamp lets the UMI-1's mic capsule plug into a 2-channel interface, so you can use loopback.

3.3 #15 To measure a radio's outputs, use a balanced USB interface plus an attenuator, never a headset adapter on both channels.

> Tying both channels' negatives to computer ground can trip the radio's protection. Put test leads on Hot and Cold with ground unconnected and divide the voltage to the interface's limit (his example takes ~1 V); a line-output converter ahead of an adapter also works. RARE
>
> **Sources:** DIYMA 2021-11-01 · DIYMA 2021-11-01 (attenuator)

In plain English: To measure what a radio actually sends out, he says to use a balanced USB interface plus an attenuator (a simple voltage divider that turns the signal down), never a headset adapter across both channels. The danger: a headset adapter ties both channels' negative wires to the computer's ground, which can trip the radio's protection circuit. Instead, put the test leads on each output's Hot (+) and Cold (−) wires, leave ground unconnected, and divide the voltage down to what the interface can accept; his example interface takes about 1 V. A line-output converter (a device that turns speaker-level signals into low-level ones) placed ahead of an adapter also works.

3.3 #16 Near-field measurements can't show cone breakup or door-panel contributions.

> Near field turns into far field at about 3× the driver's largest dimension; you must be that far away for cone and panel to sum. So a near-field anomaly (say at 2 kHz) is unlikely to be breakup, and sibilance or bad imaging is more likely EQ or reflections. RARE
>
> **Sources:** DIYMA 2022-03-16 · DIYMA 2017-07-30

In plain English: A near-field measurement puts the mic very close to the speaker, about an inch away. He says it can't show cone breakup (when parts of the cone flex unevenly at high frequencies) or what the door panel adds. The near field turns into the far field at about 3 times the driver's largest dimension, and you must be at least that far away for the sound from the cone and the panel to add together. (By that rule, for a 6.5-inch woofer that works out to roughly 20 inches, about half a meter.) So an odd bump in a near-field measurement, say at 2 kHz, is unlikely to be breakup. If you hear sibilance (harsh "s" sounds) or poor imaging, the cause is more likely EQ or reflections.

3.3 #17 Sanity-check odd results before blaming the speaker.

> A flat ~6 dB jump above 250 Hz between measurements may be the car cutting radio volume because it thinks a door is open; two identical tweeters measuring differently calls for a DC-resistance check first; claimed DSP "transient" differences can be settled with a null test (align, invert, sum). RARE
>
> **Sources:** DIYMA 2022-01-30 · DIYMA 2021-11-21 · DIYMA 2021-01-04

In plain English: Before blaming a speaker for a strange measurement, he checks the simple explanations first. His examples: a flat jump of about 6 dB above 250 Hz between two measurements may be the car itself turning the radio down because it thinks a door is open. If two identical tweeters measure differently, first check their DC resistance with a meter. And when

people claim one DSP handles "transients" (sudden sounds) differently from another, a null test settles it: line the two recordings up in time, flip one upside down (invert it), and add them. If they cancel, they are the same.

3.3 #18 "The RTA doesn't lie," but you can misread it.

> A calibrated mic is correct; the screen misleads through single-mic combing, distortion it can't show, and stereo noise with no center. Cardboard near the mic only "invalidates" a measurement if you EQ its comb without understanding it, so always ask whether a measurement makes sense.
>
> **Sources:** Mobile Electronics 2017 (tag page) · DIYMA 2017-10-28 · DIYMA 2017-06-28

In plain English: His view is that "the RTA doesn't lie": a calibrated mic tells the truth about the sound at that spot. The danger is misreading the screen. Three ways it misleads: comb-filter peaks and dips that only exist at that one mic position; distortion, which the RTA can't show; and stereo noise, which has no center. A piece of cardboard near the mic only "invalidates" a measurement if you EQ away the comb it creates without understanding it. So his habit is to always ask whether a measurement makes sense.

3.3 #19 The UMI-1 kit and binaural heads have different jobs.

> UMI-1: serialized omni mic with its own calibration file on a headrest stand, USB-Duo interface usable with REW, Open Sound Meter or SMAART, ambient noise no problem. The BHM binaural mics are for recording and comparing cars, not EQ; tuning with "ears" gains nothing and needs a new target (3.5 #9). Company copy partly.
>
> **Sources:** UMI-1v2 product page · Audiofrog binaural page · DIYMA 2021-11-23

In plain English: This contrasts two Audiofrog products. (Andy co-founded Audiofrog, and the note "Company copy partly" means some of this comes from the company's product pages rather than his own words.) The UMI-1 is a measurement kit: an omni mic with its own serial number and calibration file, a stand that mounts on the headrest, and a USB-Duo interface that works with REW, Open Sound Meter or SMAART software; background noise doesn't spoil its measurements. The BHM binaural mics are tiny mics that sit in the ears of a listener or dummy head to record what a person would hear; they are for recording and comparing cars, not for EQ. He says tuning with "ears" mics gains nothing and would need a different target curve (see 3.5 #9).

3.4 Mic placement, spatial averaging and smoothing

What this part is about: When you tune a car system, a measurement microphone stands in for your ears, and software shows how loud each pitch (frequency) is at that spot. This part covers where to put that mic, why he averages a few nearby mic positions, and how much to "smooth" (blur) the graph so you see what you actually hear.

3.4 #1 Measure from outside the car with the door shut; the target assumes an empty car.

> Strap the mic to the headrest at the listening position, run the extension cable out and close the door. Insisting on exact listening conditions "doesn't make any sense" (a chair maker doesn't need you in the chair); it also spares you 20–60 minutes of pink noise and keeps the laptop from blocking

> speakers. RARE
>
> **Sources:** UMI-1 video @11:02 · Tuning PDF v1 (~2018)

In plain English: To tune, you play test sounds and a mic at your listening spot reports what arrives there. He says: strap the mic to the headrest where your head would be, run its extension cable out of the car, close the door and run the measurement from outside. Some people insist on measuring under exact listening conditions, with themselves in the seat. He says that "doesn't make any sense", comparing it to a chair maker who doesn't need you sitting in the chair to build it. The target curve (the shape you tune toward) assumes an empty car. Measuring from outside also spares you 20–60 minutes of pink noise (a hiss-like test signal), and keeps the laptop out of the car where it could block speakers. Many people believe the "right" way is to measure with yourself in the seat; he disagrees.

3.4 #2 Don't sit in the seat waving the mic around your head.

> Your head blocks short wavelengths, so the far-side mic reads less treble, the average shows missing highs, and you add needless HF boost; results also vary day to day, and no published target (his included) assumes a body in the car. If you must, wave from the back seat, or measure the difference your body makes and fold it into the target. RARE
>
> **Sources:** DIYMA 2021-09-02 · DIYMA 2020-10-08

In plain English: Some tuners sit in the seat and move the mic around their head while measuring. He says don't. High notes (treble) have short wavelengths, and a head is big enough to block them. So whenever the mic is on the far side of your head from a speaker, it reads less treble. The averaged result then shows highs that seem to be missing, and you add high-frequency (HF) boost you don't need. Results also vary from day to day, and no published target curve, including his, assumes a person in the car. If you must do it, he says to wave the mic from the back seat, or measure how much difference your body makes and build that difference into your target.

3.4 #3 What moving the mic does, by frequency.

> His REW test (5 positions inside a 7-in circle at the driver's head): below about 200–220 Hz the curves are nearly identical; 200 Hz–1 kHz varies a lot but in broad (low-Q) features; above 1 kHz a small move changes narrow (high-Q) features by 25 dB or more.
>
> **Sources:** Audiofrog blog Jan 2021 (single vs multi-mic) · DIYMA 2016

In plain English: Hz (hertz) measures pitch: low numbers are bass, high numbers are treble, and 1 kHz is 1,000 Hz. He tested what happens when the mic moves a little. Using REW (Room EQ Wizard, a free measurement program), he measured 5 spots inside a 7-inch (about 18 cm) circle at the driver's head. Below about 200–220 Hz the 5 curves were nearly identical. From 200 Hz to 1 kHz they varied a lot, but in broad, gentle bumps ("low-Q" means wide). Above 1 kHz, a small move changed narrow, sharp features ("high-Q") by 25 dB or more. A dB (decibel) is the unit of loudness difference, and 25 dB is a very large change. The lesson: the higher the pitch, the more an inch or two of mic movement changes the fine detail on the graph.

3.4 #4 Narrow peaks and dips above about 1 kHz are a comb filter you don't hear; leave them.

> They come from two or more sources (L and R, two drivers, a speaker and its reflections) arriving at different times; head movement averages them, so "We don't hear them unless we play a sine wave." Any speaker above ~100 Hz measured at one point "will measure like crap." RARE
>
> **Sources:** Audiofrog blog Jan 2021 (single vs multi-mic) · DIYMA 2021-08-06 · DIYMA 2021-08-26

In plain English: A comb filter is what you get when the same sound reaches one spot two or more times, slightly out of step. At some pitches the copies add up (peaks); at others they cancel (dips). On a graph it looks like the teeth of a comb. He says in a car the copies come from left and right speakers, two drivers playing the same notes, or a speaker and its own reflections off glass and panels. Because your head is always moving slightly, your hearing averages these out: "We don't hear them unless we play a sine wave" (a single pure tone). He adds that any speaker above about 100 Hz measured at one single point "will measure like crap." So in practice: leave those narrow peaks and dips above about 1 kHz alone. Many people believe every peak and dip on the graph should be EQ'd away; he disagrees.

3.4 #5 One mic with 1/6-octave smoothing is good enough for production tuning.

> A spatial average correlates better with hearing, but after MS-8 Harman found a smoothed single-mic ("spectral") average about 90% as accurate, so the MS-2 used one mic; his 2021 test matched the 5-position average closely (exceptions near 2.6 kHz, 1.2 kHz, 602, 445, 334 Hz), and one live mic lets you EQ in real time. RARE **Conflict:** in 2013 he called one mic worse than a corrected multi-mic average
>
> **Sources:** Audiofrog blog Jan 2021 (single vs multi-mic) · DIYMA 2017-06-26 · DIYMA 2013

In plain English: Smoothing blurs the graph so you see the trend instead of every wiggle; 1/6-octave smoothing averages each point over a band one-sixth of an octave wide (an octave is a doubling of pitch, such as 500 to 1,000 Hz). A spatial average combines measurements from several mic spots. He agrees a spatial average matches hearing better. But after the MS-8 (a Harman/JBL car processor that used a head-worn two-mic headset), Harman found a smoothed single-mic average (a "spectral" average, meaning averaged across frequency) was about 90% as accurate, so the later MS-2 used one mic. His own 2021 test matched the 5-position average closely, except near 2.6 kHz, 1.2 kHz, 602, 445 and 334 Hz. One live mic also lets you change EQ and watch the result in real time. The conflict: in 2013 he wrote that a single calibrated mic is worse than a spatial average of several known, corrected mics. Here he says one smoothed mic is good enough for production tuning. Both statements are his, from different years.

3.4 #6 A spatial average is for high-frequency magnitude only; below about 1 kHz use a single point.

> An average throws away phase, and phase matters far more below 1 kHz, where long wavelengths make every position look alike anyway; above that, the average represents hearing better. A single mic is fine below about 800 Hz. RARE
>
> **Sources:** DIYMA 2017-06-28 · DIYMA 2016 · DIYMA 2017-10-30

In plain English: Magnitude means how loud each frequency is. Phase means timing: where each wave is in its up-and-down cycle. Phase decides whether two speakers add together or cancel. An average of several mic positions keeps the loudness but throws away phase. He says phase matters far more below 1 kHz, and down there the waves are long (about a foot at 1 kHz, and many feet in

the bass), so every position looks alike anyway and averaging gains nothing. Above 1 kHz, the average represents what you hear better. He also says a single mic is fine below about 800 Hz. So in practice: use a spatial average only for loudness above about 1 kHz, and use a single mic position below that and for anything involving phase. Many people treat the average as the best measurement for everything; he limits it to the highs.

3.4 #7 Five or six positions are enough; hundreds add nothing.

> Rotate the mic by hand around a ~7-in circle at the head, one measurement per spot; 15–20, let alone 1,000, are unnecessary. Measuring spots where nobody listens (windshield, A-pillar, dash) only dilutes the seats that matter. RARE
>
> **Sources:** DIYMA 2021-08-06 · DIYMA 2016 · DIYMA 2017-06-28

In plain English: He moves the mic by hand around a circle of about 7 inches (18 cm) at head position, taking one measurement at each spot. Five or six spots are enough; 15–20, let alone 1,000, are unnecessary. Some methods also measure spots where nobody's ears are, such as near the windshield, the A-pillar (the post on each side of the windshield) or the dash. He says that only dilutes the data for the seats that matter. Many people assume more mic positions always means a more accurate result; he says the gain stops after a handful.

3.4 #8 Average magnitudes; never sum mics or average phase.

> Wiring mics together or mixing a multi-input interface gives a phase-dependent sum (a delay-and-sum array is a beam-former), not an average; switch quickly between mics or average afterwards (SMAART, SysTune, REW). Averaging phase is meaningless: speakers at −180°/0° and at −90°/+90° both null. RARE
>
> **Sources:** DIYMA 2020-10-08 · DIYMA 2021-08-17 · DIYMA 2017-06-27

In plain English: Some people try to make a spatial average by wiring several mics together or mixing them in a multi-input audio interface. He says that gives a sum, not an average, and the sum depends on phase: where the mic signals are out of step, they partly cancel. Mics delayed and added together form a "beam-former" (a mic array that listens hardest in one direction), not an average of what your head would hear at each spot. The right way is to switch quickly between mics, or measure each one and average afterwards in software (SMAART, SysTune, REW). Averaging phase is also meaningless. His example: speakers at −180° and 0° average to −90°; speakers at −90° and +90° average to 0°. The averages differ, yet both pairs are exactly half a cycle (180°) apart, so both cancel (null). The averaged number hides that.

3.4 #9 Two practical ways to tune to an average.

> Either average six responses in REW's "All SPL" view and tune that (slow, not real time), or find the one mic position whose response best matches the average and tune live there. Either way, don't boost nulls; REW's Auto EQ can then fit the filters. RARE
>
> **Sources:** DIYMA 2021-08-09

In plain English: He gives two ways. First: measure six positions in REW, view them together in its "All SPL" view (SPL means sound pressure level, i.e. loudness), average them, and tune to that averaged curve. This is slow because you can't watch changes live. Second: find the single mic

position whose response best matches the average, leave the mic there, and tune live. Either way, don't boost nulls (deep dips caused by cancellation). Then REW's Auto EQ (a feature that calculates EQ settings for you) can work out the filters.

3.4 #10 Move the mic before you fix a narrow dip or ripple.

> Measure while slowly moving the mic (his example: a 3 kHz dip), or take a few readings around a 7-in circle: if the feature persists, fixing it may help; if it vanishes, fixing it probably makes things worse. Precision there may be "a nothingburger with extra tomatoes." RARE
>
> **Sources:** DIYMA 2011 · DIYMA 2021-03-26 · DIYMA 2021-05-07

In plain English: Before you EQ a narrow dip or a wavy ripple, check whether it's real for your ears or just an accident of that exact mic spot. Measure while slowly moving the mic (his example was a dip at 3 kHz), or take a few readings around a 7-inch (18 cm) circle. If the feature stays, fixing it may help. If it vanishes, fixing it probably makes things worse. He jokes that being precise about it may be "a nothingburger with extra tomatoes", meaning a lot of fuss over nothing. Many people EQ whatever one measurement shows; he says test it first.

3.4 #11 Smoothing: coarser up high, finer down low; zoom in only to find a cause.

> His numbers vary by year: 1/3 octave above 3 kHz, 1/6 from 800 Hz–3 kHz, 1/12 below (2017); 1/3 above 1 kHz, 1/6 below (2020); 1/3–1/6 high and 1/6–1/12 low (2022, see #12). Read an L+R sum at 1/3 first. Example: 1/3 octave showed a 6 dB rise at 300–600 Hz that 1/12 revealed as a 20 dB peak at 500 Hz plus a dip. **Era:** numbers shifted, rule didn't
>
> **Sources:** DIYMA 2017-06-26 · DIYMA 2020-07-08 · DIYMA 2012–13

In plain English: Smoothing blurs the graph. Coarse smoothing (1/3 octave) blurs a lot; fine smoothing (1/12 octave) shows more detail. His rule: coarse for the highs, fine for the lows, and zoom in only when you are hunting for the cause of a problem. His exact numbers changed by year. 2017: 1/3 octave above 3 kHz, 1/6 from 800 Hz to 3 kHz, 1/12 below. 2020: 1/3 above 1 kHz, 1/6 below. 2022: 1/3 to 1/6 for the highs and 1/6 to 1/12 for the lows. He also says to first read left and right playing together (L+R) at 1/3 octave. His example: at 1/3 octave a 6 dB rise appeared across 300–600 Hz; at 1/12 it turned out to be a 20 dB peak at 500 Hz plus a dip. The numbers moved over the years; the rule didn't.

3.4 #12 Never tune at 1/24 or 1/48 octave, and don't match L and R that finely.

> Very high resolution is for studying a speaker apart from the room (anechoic or gated), not for EQ; in a car you get lost chasing an inaudible comb. RARE
>
> **Sources:** DIYMA 2022-03-16 · DIYMA 2021-08-26

In plain English: 1/24 and 1/48 octave are extremely fine settings that show every tiny wiggle. He says that much detail is for studying a speaker on its own, apart from the room: in an anechoic chamber (a room whose walls absorb echoes) or with a "gated" measurement (one that stops recording before reflections arrive). It is not for EQ. In a car you get lost chasing comb filtering you can't hear. He also says not to match left and right that finely. Many people assume more detail always means a better tune; he disagrees.

3.4 #13 The in-car RTA sums direct and reflected sound, so a "flat" reading can sound wrong.

> Off-axis reflections can carry a dip near a crossover; EQing the combined reading flat makes the direct sound bright and fatiguing. Below 1 kHz an out-of-phase reflection can pin the image to a speaker even when the RTA matches (why far-off pillar pods cause headaches). RARE
>
> **Sources:** Tuning PDF v2 (~2023) · DIYMA 2022-02-23

In plain English: An RTA (real-time analyzer) shows how loud each frequency is at the mic. In a car, the mic hears the direct sound from a speaker plus its reflections off glass, dash and seats, all mixed together, and it can't tell them apart. He names two problems. First, reflections of sound that left the speaker off to the side (off-axis) can carry a dip near the crossover (the point where one driver hands the sound to another). If you EQ the combined reading flat, you also boost the direct sound, which makes it bright and fatiguing. Second, below 1 kHz, a reflection that arrives out of phase with a speaker's direct sound can "pin" the image (where the sound seems to come from) to that speaker, even when the RTA shows left and right matching. He says this is why pillar pods (speaker housings on the windshield posts) placed far from the nearby surfaces cause tuning headaches: the farther a speaker is from a surface, the lower in pitch that combing begins.

3.4 #14 Measure at normal listening level, and only where you'll listen.

> Measuring too loud degrades results (the MS-8 calibrated near conversation level); a mic left in the console doesn't verify a driver-seat calibration, since they are two different curves.
>
> **Sources:** Tuning PDF v1 (~2018) · DIYMA 2011

In plain English: Measuring too loud makes the results worse. He notes the MS-8 ran its calibration at about conversation level. Also, measure only where people will actually listen. A mic left sitting in the center console can't check a calibration done for the driver's seat, because the sound at those two spots is different: they are two different curves.

3.5 Target curves

What this part is about: A target curve is the shape you tune toward: how loud the bass, midrange and treble should be compared with each other at your seat. This part covers his target, where it came from, why it isn't flat, and how to use it sensibly.

3.5 #1 The Audiofrog target: about +10 dB of bass below ~60 Hz, a transition to flat midrange between ~60 and 160 Hz, and a gentle treble tilt to −6 dB at 20 kHz.

> Use −3 dB at 20 kHz if you want more highs. Cars need roughly 6–12 dB more bass than mids and highs; he builds the lift as a 12 dB/octave low shelf at about 60 Hz, adjusted to taste.
>
> **Sources:** Tuning PDF v1 (~2018) · DIYMA 2021-03-26 · DIYMA 2017-07-05

In plain English: Audiofrog is his company. Its target has bass below about 60 Hz roughly 10 dB louder than the midrange. Between about 60 and 160 Hz the curve eases down to flat. The midrange stays flat, and then the treble tilts gently down until it is 6 dB lower at 20 kHz (the top of human hearing). If you want more highs, end the tilt at −3 dB instead. He says cars need roughly 6–12 dB more bass than mids and highs. He builds that bass lift with a "low shelf" (an EQ setting that raises everything below a chosen frequency by a set amount), placed at about 60 Hz with a 12 dB-per-octave slope and adjusted to taste.

3.5 #2 The so-called "Harman curve" began as his own installer curve.

> He used it for years in retail, brought it to Harman, where it became the MS-8 curve because no Harman curve existed yet; Harman research later "kind of confirmed" it with small preference tweaks. All good-sounding car targets share this shape, but it is "not the right shape for larger rooms." He disputes that it came from David Clark. RARE
>
> **Sources:** DIYMA 2020-10-14 · Audiofrog blog Mar 2021 (target curves)

In plain English: Harman is the large audio company (owner of JBL) where he worked. People call this target shape the "Harman curve." He says it began as his own curve from his years as a retail installer. He brought it to Harman, where it became the curve in the MS-8 processor because Harman had no curve of its own yet. Harman's research later "kind of confirmed" it, with small tweaks based on listener preferences. He says all good-sounding car targets share this shape, but it is "not the right shape for larger rooms" such as a home living room. He disputes the claim that the curve came from David Clark (of DLC Design), allowing only that Clark may have contributed to an earlier target in Harman's factory-audio division.

3.5 #3 Target-curve wars are pointless: the popular curves differ by about 2 dB.

> Normalized at 1 kHz, the Audiofrog and "Half Whitledge" curves differ by about 2 dB at 20 Hz, 1.3 dB around 200 Hz (an octave wide) and 2 dB at 20 kHz, the same three tweaks his research found. People claiming one is "MUCH better" have failed to hit either. RARE
>
> **Sources:** Audiofrog blog Mar 2021 (target curves)

In plain English: Enthusiasts argue online about which target curve is best. He compared his Audiofrog curve with another popular one, the "Half Whitledge" curve, lined up so both read the same at 1 kHz. They differ by about 2 dB at 20 Hz, 1.3 dB around 200 Hz (over a band about an octave wide) and 2 dB at 20 kHz. Those are small differences, and he notes they are the same three tweaks his research had found. His conclusion: people who claim one curve is "MUCH better" have failed to hit either. Many people believe picking the right target is the key to good sound; he says the popular choices are nearly the same.

3.5 #4 The numbers drifted over the years; the shape never did.

> 2008: +6 to +9 dB below 60 Hz, flat 160 Hz to ~3 kHz, then about −3 dB/octave. 2009: highs falling gently from ~1 kHz. MS-8 (2011): +9 dB bass, flat 160 Hz–1 kHz, −6 dB at 20 kHz. 2018 PDF: +10 dB. He says the curve "has made my customers happy for 25 years" and pleases ~90% of listeners. Changed
>
> view: numbers only
>
> **Sources:** DIYMA 2008 · DIYMA 2011 · DIYMA 2010

In plain English: Over the years he quoted slightly different numbers for his target, but the overall shape stayed the same. 2008: bass 6 to 9 dB up below 60 Hz, flat from 160 Hz to about 3 kHz, then falling about 3 dB per octave (each doubling of frequency). 2009: highs falling gently from about 1 kHz. MS-8 (2011): 9 dB of bass boost, flat from 160 Hz to 1 kHz, and 6 dB down at 20 kHz. 2018 tuning PDF: 10 dB of bass. He says the curve "has made my customers happy for 25 years" and pleases about 90% of listeners. The [Changed view] tag applies to the numbers only, not to the idea.

3.5 #5 Why the treble tilts down instead of staying flat: an in-car measurement is essentially the speaker's power response.

> The mic sums direct and reflected sound, so the target is what a room-friendly speaker's power response looks like, plus bass lift (he cites Toole and Harman's speaker-and-room work). EQing that sum flat makes the direct sound too bright; with LR4 between mid and tweeter the top usually needs only a slight tilt. RARE
>
> **Sources:** DIYMA 2009 · Tuning PDF v2 (~2023) · Audiofrog blog Jan 2021 (crossovers)

In plain English: A speaker's "power response" is its total sound output in every direction, not just straight ahead. In a car, the mic picks up direct sound plus many reflections at once, so he says what it measures is essentially the power response. His target is what the power response of a speaker designed to sound good in rooms looks like, plus extra bass. He cites Floyd Toole's and Harman's research on speakers and rooms. If you EQ that combined measurement flat, the direct sound ends up too bright. He adds that with an LR4 crossover between the midrange and the tweeter (a common, fairly steep crossover type), the top end usually needs only a slight downward tilt to meet the target. Many people believe a perfect system should measure flat; he says not in a car.

3.5 #6 Start the bass rise around 160–200 Hz, not 350 Hz, and roll the top off gradually from about 2 kHz.

> A rise starting at 350 Hz sounds thin; about 5 dB too much above 2 kHz sounds bright but not piercing. Harman's early in-car tests (Sean Olive's group, 2009) preferred gradual attenuation above about 2 kHz over a longer slope starting lower. RARE
>
> **Sources:** DIYMA 2013 · DIYMA 2009

In plain English: Two details of the shape. Bass: start the rise around 160–200 Hz. He says a rise starting as high as 350 Hz sounds thin (the source doesn't spell out why). Treble: roll off gradually from about 2 kHz. About 5 dB too much treble above 2 kHz sounds bright, but not piercing. He cites Harman's early in-car listening tests (Sean Olive's research group, 2009): listeners preferred a gradual drop above about 2 kHz over a longer slope that started lower.

3.5 #7 The target's low end is also the design spec for his subwoofers.

> Audiofrog sub manuals show a modeled in-car response averaged over about 30 cars, and in the recommended box it closely matches the target's low end (unless the car is a Smart ForTwo or a school bus). He starts from the in-car target and designs the woofer for a sensible box, which is why textbook Qtc 0.707 rules don't apply. RARE
>
> **Sources:** DIYMA 2021-09-04 · DIYMA 2021-08-26

In plain English: How much bass a subwoofer makes depends on its box and on the car it's in. Audiofrog's sub manuals show a modeled in-car response averaged over about 30 cars. In the recommended box, that response closely matches the low end of his target (unless the car is a Smart ForTwo or a school bus, i.e. extremely small or extremely large). His approach is to start from the in-car target and design the woofer to suit a sensible box. That, he says, is why textbook "Qtc 0.707" rules don't apply. Qtc is a number describing how a woofer behaves in a sealed box;

0.707 is the textbook value for the flattest bass measured outside a car. Many people believe a sub should be designed and boxed to that textbook value; he designs for the response inside the car instead.

3.5 #8 Preferences change the target, never the process.

> The target is just where the objective part ends; the best one is the one that matches your (or the customer's) taste most often. A wider bass-to-mid transition (to 300 Hz or beyond) adds impact at some cost to midrange detail; a bigger bass boost makes bass-in-front harder to achieve.
>
> **Sources:** Tuning PDF v1 (~2018) · DIYMA 2020-10-03

In plain English: He says the objective, measurable part of tuning ends at the target; choosing the target is a matter of taste. The best target is the one that matches your taste (or your customer's) most often. A different preference changes the target, never the process. Two trade-offs he gives: stretching the bass-to-midrange transition up to 300 Hz or beyond adds impact, at some cost to midrange detail. A bigger bass boost makes "bass in front" harder to achieve, meaning bass that seems to come from the front of the car, where the music's stage is, rather than from the sub behind you.

3.5 #9 Head-worn and binaural mics need their own target with more treble roll-off.

> Flat on a head mic is "way too bright"; binaural and spatial-array data differ mainly in HF roll-off above about 2.5 kHz, from head masking. Adding EQ peaks and dips for "HRTF" to raise the stage misunderstands HRTF, which is about moving heads and moving sounds. RARE
>
> **Sources:** DIYMA 2013 · DIYMA 2017-06-26 · DIYMA 2020-10-14

In plain English: A binaural or head-worn mic sits at your ears, so your head shapes what it picks up. He says these mics need their own target with more treble roll-off: tuning flat with a head mic is "way too bright." Binaural data and spatial-array data (from several free-standing mic positions around the head) differ mainly in the high-frequency roll-off above about 2.5 kHz, because the head blocks (masks) short wavelengths. HRTF (head-related transfer function) describes how your head and ears change sound coming from different directions. Some people add EQ peaks and dips to imitate it, hoping to raise the stage (make the music seem to come from higher up). He says that misunderstands HRTF, which is about moving heads and moving sounds.

3.5 #10 A shared target curve means nothing without the method used to measure it.

> A curve that won "best sound" at the 2013 Eurofinals has "way too much midbass" if measured with one mic and pink noise, and results also depend on tweeter directivity; always state mic, averaging and signal. RARE
>
> **Sources:** DIYMA 2016 · DIYMA 2012

In plain English: A target curve posted online means little unless you know how it was measured. His example: a curve that won "best sound" at the 2013 Eurofinals (a car-audio sound competition) has "way too much midbass" if it was measured with one mic and pink noise. The result also depends on how directional the tweeters are. So, he says, always state the mic, the averaging and the test signal you used. Many people copy a winner's curve expecting the same result; he says that doesn't work without the method.

3.5 #11 Tweeterless systems: drop the treble tilt, and let the aim decide the HF EQ.

> Full-range drivers aimed at you keep highs in the direct sound: tight, spaceless images, and the boost you add makes it bright. Off axis they need more boost and sound dull. Aimed at the windshield (as OEMs do), the glass becomes "a tweeter without having to pay for a tweeter"; still, every car sounds better with tweeters. RARE
>
> **Sources:** Tuning PDF v2 (~2023) · DIYMA 2021-12-02

In plain English: Some systems have no tweeters and use small full-range drivers for mids and highs. For these he says: drop the downward treble tilt from the target, and let the speakers' aim decide the high-frequency EQ. He explains that such drivers send their highest notes in a narrow beam. Aimed at you, the highs arrive mostly as direct sound: tight images with no sense of space, and the treble boost you add makes it bright. Aimed off to the side, they need more boost and sound dull. Aimed at the windshield, as carmakers (OEMs) do, the glass becomes "a tweeter without having to pay for a tweeter." Still, he says every car sounds better with tweeters.

3.5 #12 REW house-curve settings that approximate his target:

> LF Rise Start 160 Hz, LF Rise End 60 Hz, LF Rise Slope about 12, HF Fall Slope about 1.6 (units not stated). Testgear also supplies target files for each analyzer.
>
> **Sources:** DIYMA 2014 · Testgear site

In plain English: REW (Room EQ Wizard) lets you describe a target, which it calls a "house curve," with a few settings. He gave these to approximate his target: LF Rise Start 160 Hz, LF Rise End 60 Hz, LF Rise Slope about 12, HF Fall Slope about 1.6. LF means low frequency (bass) and HF means high frequency (treble). So the bass starts rising at 160 Hz and stops rising at 60 Hz. He didn't state the units for the two slopes. His Testgear site also supplies ready-made target files for each analyzer.

3.6 EQ: what to fix, what to leave, and how

What this part is about: EQ (equalization) means turning particular frequencies up or down to correct the sound. This part covers which problems EQ can fix, which it can't, and the habits he uses to apply it well.

3.6 #1 Separate left and right EQ is mandatory.

> You tune each side to the target from its own position; DSPs that link L/R EQ can't run his process. At Harman, a German automaker that forced identical L/R filters (citing "phase problems") got bad-sounding cars. RARE
>
> **Sources:** Tuning PDF v1 (~2018) · DIYMA 2015

In plain English: In his process, the left speakers and the right speakers are each tuned to the target separately, each measured from its own position. So a DSP (digital signal processor: the box that handles EQ, crossovers and delays) that links left and right EQ, forcing the same settings on both, can't run his process. He recalls that at Harman, a German automaker forced identical left and right filters, citing "phase problems," and got bad-sounding cars. Many people believe left and right should get identical EQ; he says each side needs its own.

3.6 #2 Matching left to right matters more than hitting the target.

> Match within a couple of dB; if one side has a dip you can't fix, put the same dip on the other side to protect the center. After matching, deviate only where needed to stop the image wandering ("The windshield is the devil for everything but a center speaker"). RARE
>
> **Sources:** Tuning PDF v2 (~2023) · DIYMA 2014

In plain English: He says left and right matching each other, within a couple of dB, matters more than both hitting the target. If one side has a dip you can't fix, put the same dip on the other side, to protect the center (keep sounds meant to be in the middle, like a singer's voice, centered). Once they match, depart from the match only where needed to stop the image wandering, meaning sounds drifting sideways as the pitch changes, which the car's shape and reflections can cause. "The windshield is the devil for everything but a center speaker," he says. Many people treat hitting the target as the main goal; he puts left-right matching first.

3.6 #3 Treat peaks as peaks: put the target at the dips and cut down to it.

> Don't read the peak tops as the speaker's level or the dips as "it won't play that low"; slide the target onto the dips like "a router template" and cut each peak by its excess (dip at 57.5 dB → target there; a 64 dB peak at 1.5 kHz → cut ~7 dB), then restore level with amp input sensitivity. You'll find more bandwidth and standard crossovers work. RARE **Conflict:** PDFs set midrange at 0 dB instead
>
> **Sources:** UMI-1 video @16:00 · DIYMA 2021-08-20 · DIYMA 2021-08-06

In plain English: A measured response is full of peaks and dips. He warns against two misreadings: treating the tops of the peaks as the speaker's true level, and reading the dips as "it won't play that low." Instead, slide the target curve down until it sits on the dips, like "a router template" (a guide a woodworker traces along), then cut each peak by however much it sticks up above the target. His example: a dip reads 57.5 dB, so the target goes there; a peak at 1.5 kHz reads 64 dB, so cut about 7 dB there. Then restore the overall loudness with the amplifier's input sensitivity (gain) setting. He says you'll find the speaker plays a wider range (more bandwidth) than you thought, and standard crossover settings work. The conflict: his tuning PDFs instead line up the target so the midrange sits at 0 dB, rather than sliding it down to the dips.

3.6 #4 An amp with built-in DSP takes that trick away.

> With no analog gain stage after the DAC, once the DSP outputs are at maximum that's all the level you get, so you must boost dips as well as cut peaks. RARE
>
> **Sources:** DIYMA 2022-03-03

In plain English: Some amplifiers have a DSP built in. In those, there is no analog gain stage after the DAC (the chip that turns the digital signal back into an audio signal), so there is no amplifier gain to turn up afterwards. Once the DSP outputs are at maximum, that's all the level you get. So with these amps you must boost the dips as well as cut the peaks, instead of only cutting as in the previous point.

3.6 #5 Use channel levels for broad regions first; EQ only what's left.

> Raise a channel a little for more 80 Hz, then remove remaining peaks. Boost on a band plus lower

> channel gain is the same as a cut.
>
> **Sources:** Tuning PDF v1 (~2018) · DIYMA 2022-03-16 · DIYMA 2015

In plain English: Each speaker channel has its own overall level. He says to use those channel levels to fix broad regions first, then use EQ only for what's left. His example: raise a channel a little to get more 80 Hz, then remove the peaks that remain. He also notes that boosting one EQ band while lowering that channel's overall gain is the same as a cut, because the response ends up with the same shape.

3.6 #6 Compute a parametric filter's Q from the peak's −3 dB bandwidth.

> Q = center frequency ÷ the width between the points 3 dB below the peak, read at 1/12 octave. Worked example: peak at 474 Hz, −3 dB at 398 and 571 Hz → Q 2.73, refined by ear to −12 dB at Q 1.9. (The energy definition, 2πf × stored ÷ lost, gives similar, not identical, values.)
>
> **Sources:** Tuning PDF v2 (~2023) · DIYMA 2017-09-02 · DIYMA 2021-12-22

In plain English: A parametric EQ filter has three settings: its center frequency, its gain (how much it boosts or cuts) and its Q (how narrow it is; higher Q means narrower). He shows how to work out Q from a measurement read at 1/12-octave smoothing. Find the peak's center frequency, then the two points on either side where the response is 3 dB below the top. Q = center frequency ÷ the width between those two points. His example: a peak at 474 Hz with −3 dB points at 398 and 571 Hz, so Q = 474 ÷ (571 − 398), which he gives as 2.73. He then refined it by ear to a 12 dB cut at Q 1.9. He notes the energy definition of Q (2π times frequency, times energy stored, divided by energy lost) gives similar, but not identical, values.

3.6 #7 Keep EQ broad, especially above 1 kHz.

> Don't chase narrow, position-dependent features; avoid parametric Q above about 6. Treble cuts are fine when wide (his examples: −4.5 dB at 10 kHz and −4 dB at 15 kHz, both Q 1.4). **Conflict:** the PDFs, even v2, still show a Q 10 cut at 1.6 kHz
>
> **Sources:** Audiofrog blog Jan 2021 (single vs multi-mic) · Tuning PDF v2 (~2023)

In plain English: Higher Q means a narrower filter. He says to keep EQ broad, especially above 1 kHz, where narrow features change with every small move of the head (see 3.4). Don't chase those narrow, position-dependent features, and avoid parametric Q above about 6. Cutting treble is fine when the cut is wide: his examples are −4.5 dB at 10 kHz and −4 dB at 15 kHz, both at Q 1.4. The conflict: his tuning PDFs, even the 2023 version, still show a narrow Q 10 cut at 1.6 kHz.

3.6 #8 Cut rather than boost; near a crossover boost the low-passed driver, not the high-passed one.

> Boosting the high-pass driver near its crossover overstresses it, and boosting in or near a crossover hole is often a bad idea. Bass boosts cause most clipping: after the crossover it shows in one channel, before it (or at the A/D) in all. Tweeter crackle that appears when you turn the sub up is digital clipping from too much boost. **RARE**
>
> **Sources:** Tuning PDF v1 (~2018) · DIYMA 2021-07-08 · DIYMA 2021-11-14

In plain English: He prefers cutting peaks to boosting dips. A crossover splits the music between drivers: a high-pass filter sends only higher notes to a smaller driver such as a tweeter, and a low-pass filter sends only lower notes to a bigger one. If you must boost near the crossover point, boost the low-passed driver, not the high-passed one; boosting the high-passed driver near its crossover overstresses it. Boosting in or near a crossover hole (a dip where the two drivers don't add up well) is often a bad idea. Bass boosts cause most clipping (harsh distortion when a signal is pushed past the most the electronics can handle). Boost applied after the crossover clips in one channel; boost before it, or at the A/D (the input's analog-to-digital converter), clips in all channels. Tweeter crackle that appears when you turn the sub up is digital clipping from too much boost.

3.6 #9 Never significantly boost a null; the nicer graph is "PROOF that reflected sound matters."

> A 12 dB boost into a null clips the amp and overdrives the speaker; the mic improves only because extra reflected energy fills the dip, and it sounds "nasty." Leave a −6 dB cancellation dip alone. RARE
>
> **Sources:** DIYMA 2012 · DIYMA 2011 · DIYMA 2021-08-09

In plain English: A null is a deep dip where sound waves cancel, for example a speaker's direct sound and a reflection arriving out of step. He says a 12 dB boost into a null clips the amp and overdrives the speaker. If the graph then looks better, it's only because extra reflected energy fills the dip at the mic: that nicer graph is "PROOF that reflected sound matters," and it sounds "nasty." Leave a −6 dB cancellation dip alone. Many people boost any dip the analyzer shows; he says never boost these significantly.

3.6 #10 "Does it respond to EQ?" sorts the three kinds of dip.

> A box or driver peak is minimum phase, so EQ fixes magnitude and phase together (car systems are often close enough). A dip from a driver's own reflection can't be EQ'd. A dip that appears only with both sides playing is two speakers interacting: a phase or delay fix, not EQ. RARE
>
> **Sources:** DIYMA 2021-11-02 · DIYMA 2016

In plain English: He sorts dips with one question: does it respond to EQ? First kind: a peak from the speaker box or the driver itself. That is "minimum phase," a technical term meaning its loudness and timing errors are tied together, so EQ fixes both at once, and he says car systems often behave close enough to this. Second kind: a dip caused by a driver's own reflection, which EQ can't fix. Third kind: a dip that appears only when both sides play together. That is the two speakers interacting, and it needs a phase or delay fix, not EQ.

3.6 #11 Find the surface behind a reflection null with path arithmetic, then accept it.

> The null sits where the reflected path is half a wavelength longer: about 11 in extra for 600 Hz; a surface 1 ft away gives ~280 Hz, 6 in ~550 Hz. One reflection makes a deep narrow dip, many make wide shallow ones (more audible). A 300–500 Hz dip is usually the door glass, the price of width; "It's always something." RARE
>
> **Sources:** DIYMA 2021-11-06 · Audiofrog blog Jun 2021 (dispersion and reflection) · Audiofrog blog Dec 2020 (Time Alignment Part 2)

In plain English: A reflection travels a longer path than the direct sound. When the extra distance equals half a wavelength, the reflection arrives exactly out of step and cancels that frequency, making a null. He uses that to work backward to the surface causing it: about 11 inches (28 cm) of extra path for 600 Hz; a surface 1 foot (30 cm) away gives a null near 280 Hz, and one 6 inches (15 cm) away near 550 Hz. One reflection makes a deep, narrow dip; many reflections make wide, shallow dips, which are more audible. A 300–500 Hz dip is usually the door glass. He calls it the price of width: elsewhere he explains that mids low in the doors give a wide stage, and the side-glass dip is the trade-off. So you accept it. "It's always something."

3.6 #12 Aiming isn't an EQ substitute, but in a 2-way the aim decides whether to EQ the crossover dip.

> A 6-in is near-omni below ~500 Hz, a 2-in below ~2.7 kHz, a 1-in tweeter's aim matters only above ~10 kHz. Aimed on axis, the 2-way's hole is in the reflections: ignore it. Aimed off axis, it's in the direct sound: you may EQ it, at the passenger's expense. A 3-way removes the trade. RARE
>
> **Sources:** Audiofrog blog Jun 2021 (dispersion and reflection) · DIYMA 2021-11-30

In plain English: Aiming a speaker doesn't replace EQ. But in a 2-way system (a woofer plus a tweeter on each side), the aim decides whether to EQ the dip at the crossover. Speakers spread sound almost evenly in all directions (near-omni) at low frequencies and beam at high ones. His figures: a 6-inch (15 cm) driver is near-omni below about 500 Hz, a 2-inch (5 cm) below about 2.7 kHz, and a 1-inch (25 mm) tweeter's aim matters only above about 10 kHz. Aimed at you (on axis), the 2-way's crossover hole is in the reflections, so ignore it. Aimed away (off axis), the hole is in the direct sound, so you may EQ it, at the passenger's expense. A 3-way (adding a midrange between the two) removes that trade-off.

3.6 #13 Dash speakers have too much bass, not too little treble: EQ by cutting.

> The dash adds about 6 dB and the windshield about 6 dB more at low frequencies (depending on rake); cutting it lowers power, excursion and distortion. Don't boost the natural HF roll-off. A vent-mounted mid's peaks just need turning down, then the tweeter can cross at 2.5 kHz LR4. RARE
>
> **Sources:** Audiofrog blog Jun 2021 (dispersion and reflection) · DIYMA 2021-09-20

In plain English: Speakers in the dash get extra bass from the surfaces around them. He says the dash adds about 6 dB, and the windshield about 6 dB more, at low frequencies (how much depends on the windshield's angle, or "rake"). So a dash speaker has too much bass, not too little treble, and the fix is to cut the excess. Cutting also lowers the power needed, the cone's movement (excursion) and distortion. Don't boost the speaker's natural treble roll-off. A midrange mounted in an air-conditioning vent just needs its peaks turned down; then the tweeter can cross at 2.5 kHz with an LR4 crossover. Many people would boost the treble of a dash speaker; he says cut the bass instead.

3.6 #14 A 250 Hz "resonance" is usually a reflection peak to EQ, not a panel to deaden.

> Play a 250 Hz tone: no buzz or rattle means constructive interference from a cabin dimension, so just cut it. RARE
>
> **Sources:** DIYMA 2021-09-07

In plain English: Many people blame a peak around 250 Hz on a vibrating panel and add sound-deadening material (mats that stop panels vibrating). He says it is usually a reflection peak instead. His test: play a 250 Hz tone. If nothing buzzes or rattles, it's constructive interference (sound waves adding up) caused by one of the cabin's dimensions, so just cut it with EQ.

3.6 #15 A 60–80 Hz dip in one midbass is usually a cross-car mode: cross above it to the sub instead of EQing.

> In a thread about a subwoofer dip he traced the same kind of dip to two paths (sub→mic and sub→far side→mic) differing by ~7 ft; moving the mic to the other seat also helps confirm it. "Crossing midbass at 40 Hz isn't the way to get great midbass." RARE
>
> **Sources:** DIYMA 2022-03-16 · DIYMA 2021-12-02

In plain English: A "mode" is a frequency where sound bouncing between two surfaces reinforces or cancels itself; a cross-car mode forms across the width of the cabin. He says a 60–80 Hz dip in only one midbass (the driver that plays upper bass, usually in a door) is usually this. Instead of EQing it, set the crossover above the dip and let the subwoofer play that range. To confirm (he gave this test in a thread about a similar dip from a subwoofer): compare the distance from the sub to the mic with the distance from the sub to the far side of the car and on to the mic; a difference of about 7 feet (2.1 m) points to this mode. Or move the mic to the other seat. His view: "Crossing midbass at 40 Hz isn't the way to get great midbass."

3.6 #16 A ~2 kHz dip in a dash center channel is fine, even helpful.

> Ear spacing puts a dip near 2 kHz in the familiar stereo phantom center, so a center with that windshield dip sounds natural; L/C/R "voicing" matters less because you EQ anyway. RARE
>
> **Sources:** DIYMA 2021-09-22

In plain English: A center channel is a speaker in the middle of the dash. In the dash it may show a dip near 2 kHz from its reflection off the windshield. He says that's fine, even helpful. With ordinary stereo, a voice in the middle is a "phantom center" created by the left and right speakers together, and the spacing of your ears already puts a dip near 2 kHz in it. So a center speaker with that windshield dip sounds natural and familiar. He adds that matching the "voicing" (tonal character) of left, center and right speakers matters less in a car, because you EQ them anyway.

3.6 #17 The usual car faults: a broad peak near 600 Hz and too much midbass/lower midrange.

> The 600 Hz peak makes a car sound loud; because midbass is "always too loud," a correct tune often sounds quieter. In-car sub graphs include cabin gain, so cut a boomy peak 6–10 dB, not the 15 dB shown. A real customer tune: cut a little near 1 kHz and between 50–80 Hz. RARE
>
> **Sources:** DIYMA MS-8 FAQ · DIYMA 2015 · DIYMA 2018-06-27

In plain English: The faults he finds most often: a broad peak near 600 Hz, and too much midbass and lower midrange. The 600 Hz peak makes a car sound loud. Because midbass is "always too loud," a correct tune often sounds quieter. "Cabin gain" is the extra low bass a small, closed space like a car adds. In-car graphs of a sub already include it, so for a boomy peak, cut 6–10 dB, not the full 15 dB the graph shows. One real tune he did for a customer: cut a little near 1 kHz and between 50–80 Hz.

3.6 #18 Make sure EQ reaches above 10 kHz; he fought for it at Harman.

> Engineers there called HF EQ invalid; he argued it improved cars and the MS-8 shipped with it. If an analyzer's suggested EQ stops at a "brick wall" at 10 kHz, its algorithm may not go that high; a far-off-axis tweeter needs boost up there. RARE
>
> **Sources:** DIYMA 2015 · DIYMA 2015 (off-axis tweeter)

In plain English: At Harman, engineers called high-frequency (HF) EQ invalid. He argued it improved cars, and the MS-8 shipped with it. If an analyzer's suggested EQ stops dead at 10 kHz, like a "brick wall," its algorithm may simply not go that high. A tweeter mounted far off axis (pointing well away from you) needs boost up there. So in practice: make sure your EQ can reach above 10 kHz.

3.6 #19 Parametric EQ for fixing, graphic EQ for drawing.

> A 31-band EQ doesn't do what its sliders show (raising every band 12 dB isn't a flat +12 dB; adjacent pairs rarely render as drawn); 1/3-octave EQ "is insufficient for any real correction," and an EQ with only ±6 dB range can't tune a car. Graphics suit shaping an already-corrected curve. RARE
>
> **Sources:** DIYMA 2009 · DIYMA 2009 (31-band) · DIYMA 2016

In plain English: A graphic EQ has fixed bands with sliders, such as 31 bands spaced 1/3 octave apart. A parametric EQ lets you set each filter's exact frequency, width (Q) and gain. He says a 31-band graphic doesn't do what its sliders show: raising every band 12 dB doesn't give a flat +12 dB, and neighboring bands set in opposite directions rarely come out as drawn. He says 1/3-octave EQ "is insufficient for any real correction," and an EQ with only ±6 dB of range can't tune a car. So use parametric EQ to fix problems; a graphic suits shaping a curve that has already been corrected. Many people tune with a 31-band graphic; he says it isn't enough for correction.

3.6 #20 Use auto-EQ, but limit each channel to its passband.

> REW's auto-EQ is "damn good" and saves time; but run against the full target on every channel it flattens your crossover slopes. Auto-EQs need phase, so they build minimum phase from the average and must cap Q and boost to avoid boosting nulls; pasting a single-mic phase into a spatial average improves the model. RARE
>
> **Sources:** DIYMA 2016 · DIYMA 2021-04-17 · DIYMA 2017-06-27

In plain English: Auto-EQ is a feature that calculates EQ filters for you to match a target. He calls REW's auto-EQ "damn good" and a time saver. But if you run it against the full target on every channel, it pushes each driver toward the target everywhere and flattens the crossover slopes you set. His explanation points to the fix in the heading: limit each channel's EQ to its passband (the range that channel is meant to play). Auto-EQs also need phase, which an average doesn't have, so they build a "minimum phase" estimate from the average. That estimate hides the regions that don't behave that way, such as cancellations, so the software must cap Q and boost to avoid boosting nulls. Pasting the phase from a single-mic measurement into a spatial average improves that model.

3.6 #21 Hiss after auto-EQ on a factory source is EQ boost, not a ground fault.

> If the head unit rolls off the highs, auto-EQ boosts the top end to compensate and you hear the boost; don't tear the install apart. RARE
>
> **Sources:** DIYMA 2021-08-08

In plain English: Some factory head units (the car's original radio) roll off the highs. Auto-EQ then boosts the top end to compensate, and you hear that boost as hiss. He says this is EQ boost, not a ground fault (a wiring problem that causes noise), so don't tear the installation apart looking for one.

3.6 #22 After tuning, fix only what a notepad and the RTA confirm.

> Note track and time whenever something annoys you; later play that passage on the RTA, then check the pink-noise curve there: a peak gets fixed, otherwise it's the recording. Cutting everything occasionally annoying ends in a "smile" curve; peaks are more audible than dips. RARE
>
> **Sources:** DIYMA 2021-08-30

In plain English: After the main tune, don't keep tweaking at random. Whenever something annoys you while listening, write down the track and the time. Later, play that passage with the RTA running, then check your pink-noise curve at the same frequencies. If there's a peak there, fix it; if not, the problem is in the recording. Cutting everything that's occasionally annoying ends in a "smile" curve (bass and treble up, midrange scooped out, like a smile drawn on the graph). He notes that peaks are more audible than dips.

3.6 #23 EQ removes linear differences: cone "sound," L/C/R matching, road noise.

> Linear timbre differences between cone materials EQ away (distortion doesn't); EQ is a fine way to match L, C and R without identical speakers. Road noise is an EQ-only change (he kept a preset for a pebbled stretch near El Paso); crossovers, delays and gains stay put.
>
> **Sources:** DIYMA 2016 · DIYMA 2021-01-02 · DIYMA MS-8 FAQ (road noise)

In plain English: "Linear" differences are differences in frequency response (how loud each pitch is), which EQ can change. Distortion means the speaker adds sounds that weren't in the signal, and EQ can't remove that. He says the timbre (tonal "sound") differences between cone materials are linear and EQ away; distortion doesn't. So EQ is a fine way to match left, center and right speakers without making them identical. Road noise is also a job for EQ only: he kept a preset for a pebbled stretch of road near El Paso. Crossovers, delays and gains stay put. Many people insist that left, center and right speakers must be identical to match; he says EQ handles it.

3.7 Crossovers: points, slopes and filter types

What this part is about: Crossovers, the filters that decide which speaker plays which range of notes. It covers where he sets them, how steep they should be, which filter types he uses and why, and the common crossover "tricks" he thinks make cars sound worse.

3.7 #1 LR4 on every crossover by default.

> Reasons: best protection for the high-passed driver (it nearly removes power at resonance), a narrow transition band (what cars need, given high power and compromised placement), a target phase difference of 0° that phase-reading analyzers can check, and acoustic slopes that follow the electrical ones when drivers are flat about an octave either side. If an amp's built-in crossover is 12 dB/oct, disable it and do LR4 in the DSP. RARE
>
> **Sources:** Tuning PDF v2 (~2023) · DIYMA 2021-09-27 · DIYMA 2020-10-03

In plain English: A crossover is a filter that splits music by pitch, so each speaker only gets the notes it can handle: highs to the tweeter, lows to the woofer. A "high-pass" filter lets highs through and blocks lows; a "low-pass" does the opposite. "LR4" is one specific recipe: Linkwitz-Riley, 4th order, which cuts at 24 dB per octave (an octave is a doubling of pitch, such as 1 kHz to 2 kHz). He uses it everywhere by default, for four reasons. It protects the high-passed speaker best, because it almost removes power at that speaker's resonance (the pitch where it naturally wants to move the most). The zone where both speakers overlap is narrow, which suits cars, with their high power and poor speaker locations. Done right, the two speakers should be in step (0° phase difference), which phase-reading analyzers can check. And if the speakers are themselves flat for about an octave either side, the sound you hear follows the filter's slopes. So in practice: if an amplifier's built-in crossover is only 12 dB/oct, switch it off and set LR4 in the DSP (the digital sound processor).

3.7 #2 Pick crossover points in this order: protect the smaller driver, cross where the larger one still disperses widely, then worry about in-car response.

> Never cross below resonance except on a sub; tweeters at about 1.5–2× Fs. In small pods Fs becomes Fc, so measure it: with a DATS, or a multimeter on current in series with the speaker during a quiet sine sweep (minimum current = resonance). RARE **Conflict:** 2009 rule of 2× Fs and ~3 kHz for 1-in tweeters
>
> **Sources:** DIYMA 2021-07-12 · DIYMA 2021-08-26 · DIYMA 2009

In plain English: He chooses crossover points in a fixed order. First, protect the smaller speaker (for example, the tweeter) from notes too low for it. Second, cross where the bigger speaker still spreads its sound widely, before it starts to "beam" (send sound mostly straight ahead). Only then does he worry about how the response looks in the car. Every speaker has a resonance frequency, called Fs, where it moves most easily. He says never cross below it, except on a subwoofer, and put tweeters at about 1.5 to 2 times their Fs. In a small pod (a small sealed housing), the trapped air raises the resonance; the new value is called Fc, so you have to measure it. Two ways: a DATS (a speaker-testing tool), or a multimeter reading current, wired in series with the speaker while a quiet sine-wave sweep plays. The pitch where the current is lowest is the resonance. The [Conflict] tag: in 2009 his rule of thumb was never below 2× Fs, and about 3 kHz for 1-inch car tweeters; his later posts allow 1.5–2× Fs.

3.7 #3 Don't play mids and midbass as low as they can stand.

> "Resonance IS distortion": at and below it the spider (his "parking brake") dominates; above it, driver distortion is a "nothingburger." Each octave lower roughly doubles power and quadruples excursion. GB25 midrange: 200 Hz if you must, 250 Hz safer, above 300 Hz a waste; a 4-in dash mid at 100 Hz only makes sense with no midbass. RARE
>
> **Sources:** DIYMA 2022-03-10 · DIYMA 2021-12-14 · DIYMA 2017-08-17

In plain English: Many people set each speaker to play as low as it can without obvious trouble. He disagrees. "Resonance IS distortion" is his phrase. At and below resonance, the spider dominates the cone's motion. The spider is the springy ring that holds the cone centred; he calls it the "parking brake". Above resonance, he says, the speaker's own distortion is a "nothingburger": not worth worrying about. Each octave lower roughly doubles the power the speaker must take and quadruples how far the cone moves (its excursion). His numbers for the Audiofrog GB25 midrange (Audiofrog is his company): 200 Hz if you must, 250 Hz is safer, and above 300 Hz is a waste. Hz (hertz) measures pitch; a lower number is a lower note. A 4-inch dash midrange crossed at 100 Hz only makes sense if there is no midbass speaker.

3.7 #4 His dispersion chart grades crossover points green, yellow, orange and red.

> A 6-in crossed to a 1-in tweeter at 5 kHz is "in the red zone"; 1.5–2 kHz is much better; adding a 2.5-in midrange crossed at 2.5 kHz lands in the green. Crossing where the larger driver has narrowed leaves a dip in the reflected sound the RTA can't separate, so it's hard to tune. Protect the tweeter first; yellow only if unavoidable, never red. RARE
>
> **Sources:** Tuning PDF v2 (~2023) · DIYMA 2020-12-23

In plain English: Dispersion is how widely a speaker spreads its sound. As notes get higher, a cone spreads its sound less, and bigger cones narrow sooner. His tuning PDF has a chart that colour-codes crossover points like a traffic light: green is good, yellow and orange are worse, red is bad. A 6-inch woofer crossed to a 1-inch tweeter at 5 kHz is "in the red zone", because the 6-inch already beams there. Crossing at 1.5–2 kHz is much better, and adding a 2.5-inch midrange crossed to the tweeter at 2.5 kHz lands in the green. Why it matters: if the bigger speaker has already narrowed, the sound bouncing off the car's surfaces has a dip near the crossover. An RTA (real-time analyzer: a microphone plus a screen showing level at each pitch) can't separate that from the direct sound, so the car becomes hard to tune. His order: protect the tweeter first; accept yellow only if you can't avoid it; never red.

3.7 #5 Cars need 3-ways more than studios do, because close reflections expose a 2-way's off-axis hole.

> A 2-way crossed where the mid beams can be flat on axis with a big hole off axis; the mic sees the hole, and boosting it puts a peak in the direct sound (piercing highs near the crossover). Studio monitors get away with 2-ways at ~3 ft with walls ~10 ft away. Home-speaker crossover rules still apply in a car and make EQ easier. RARE
>
> **Sources:** DIYMA 2021-01-11 · DIYMA 2022-01-05 · DIYMA 2013

In plain English: A 2-way system has a woofer and a tweeter on each side; a 3-way adds a midrange between them. People often ask why cars need 3-ways when many top studio monitors are 2-ways. His answer is reflections. A 2-way crossed where the woofer has started to beam can measure flat straight in front but have a big "hole" in the sound going off to the sides. In a car, surfaces are so close that the microphone picks up that hole. If you boost EQ to fill it, you also boost the direct sound, which creates a peak: piercing highs near the crossover. Studio monitors get away with 2-ways because the listener sits about 3 feet (0.9 m) away, with walls about 10 feet (3 m) away. His point: home-speaker crossover rules still apply in a car, and following them makes EQ easier.

3.7 #6 Don't lower the tweeter crossover to stop a 6-in from beaming; build a 3-way.

> Crossing every driver as low as possible wears speakers faster and can make the target harder to hit; a 6-in plus a tweeter "is not a great combo." The tweeter point can sit a little below 2× Fs only with steep slopes and careful tuning. RARE
>
> **Sources:** OFCAP Ep. 19 @1:41:04 · OFCAP Ep. 19 @1:44:09

In plain English: "Beaming" means a speaker sends its sound mostly straight ahead, which large cones do at higher notes. One tempting fix is to cross the tweeter lower so the 6-inch stops sooner. He says build a 3-way instead. Crossing every speaker as low as possible wears speakers out faster and can make the target curve (the tonal balance you aim for) harder to hit. In his words, a 6-inch plus a tweeter "is not a great combo." He allows the tweeter crossover a little below 2× Fs (twice its resonance) only with steep slopes and careful tuning.

3.7 #7 His actual crossover numbers.

> PDF 3-way: sub 60 Hz LP, midbass 80 Hz HP, mid/midbass 300 Hz, mid/tweeter 2.2 kHz. 2022 Audiofrog 3-way: sub 60 Hz; GB60 80–300 Hz; GB25 300 Hz–3 kHz; tweeter from 3 kHz (2.5 kHz with 24 dB/oct; ~3 kHz with shallower; 1.8 kHz only with LR4, and a 3-way never needs it). GB10 tweeter: 2.2 kHz+ LR4 or 2.5 kHz+ at ≥12 dB/oct. A 3-way's midrange needn't play low.
>
> **Sources:** Tuning PDF v1 (~2018) · DIYMA 2022-03-10 · DIYMA 2017-10-29

In plain English: These are the specific settings he has published. His tuning PDF's 3-way: subwoofer low-pass at 60 Hz, midbass high-pass at 80 Hz, midrange-to-midbass crossover at 300 Hz, and midrange-to-tweeter at 2.2 kHz (2,200 Hz). For an Audiofrog 3-way in 2022: sub to 60 Hz; GB60 midbass from 80 to 300 Hz; GB25 midrange from 300 Hz to 3 kHz; tweeter from 3 kHz. The tweeter can go down to 2.5 kHz with 24 dB/oct slopes, about 3 kHz with shallower slopes, and 1.8 kHz only with LR4, and he adds that a 3-way never needs that. For the GB10 tweeter: 2.2 kHz and up with LR4, or 2.5 kHz and up with a slope of 12 dB/oct or steeper. Finally, the midrange in a 3-way doesn't need to play low notes.

3.7 #8 Raising mid crossovers to cure "distortion" usually hides a different fault.

> The real cause is typically DSP input clipping, clipping inside the DSP, or a leaky pod. RARE
>
> **Sources:** DIYMA 2022-03-10

In plain English: When a midrange or midbass sounds distorted, a common reaction is to raise its crossover so it plays fewer low notes. He says that usually just hides a different fault. Clipping is what happens when a signal is too big for the electronics: its peaks get chopped off, which sounds harsh. The usual real causes he names: clipping at the DSP's input, clipping inside the DSP, or a leaky pod (air escaping from the speaker's housing).

3.7 #9 Set the sub low-pass half an octave below the midbass high-pass (60 Hz LP, 80 Hz HP, both LR4).

> Most people raise the sub 9–12 dB to sound right; with 80/80, that lifts the sub's real acoustic crossover and puts sub and front out of phase, so people misuse delay to drag bass forward. Rule: drop the sub low-pass half an octave per 12 dB of extra sub level, or lift the bass with a 12 dB/oct low shelf at ~60 Hz. RARE **Changed view:** MS-8 era used the same point, 80/80
>
> **Sources:** DIYMA 2020-12-30 · DIYMA 2021-08-21 · DIYMA 2015

In plain English: The sub's low-pass (where it stops playing upward) and the midbass's high-pass (where it stops playing downward) are often set to the same number. He sets them about half an octave apart: sub low-pass 60 Hz, midbass high-pass 80 Hz, both LR4. His reason: most people turn the sub up 9–12 dB so bass sounds right (dB, decibels, measure loudness; 10 dB more sounds roughly twice as loud). With both at 80 Hz, that boost moves the sub's real, acoustic crossover higher, and the sub and front speakers end up out of phase (one pushing while the other pulls, so they work against each other). People then misuse delay to drag the bass forward. His rule: lower the sub low-pass by half an octave for every 12 dB of extra sub level, or instead lift the bass with a low shelf (an EQ setting that raises everything below a chosen pitch) at about 60 Hz with a 12 dB/oct slope. The [Changed view] tag: in the MS-8 era (a processor he worked on at Harman) he used the same point for both, 80/80.

3.7 #10 Bass "in front" is sub–midbass phase at the crossover, not a low or big midbass.

> Wrong high-pass/low-pass phase is the #1 reason bass won't come from the front, and delay can't fix it (the half-wavelength-delay-plus-flip trick works only at one frequency). Aim to stay far from 180°, not to hit exactly 0° (60° still sums louder); check by proxy: measured slopes and −6 dB point match the target, delays correct. "Impact" lives near the mid/tweeter crossover (~3 kHz), not at 40 Hz. RARE
>
> **Sources:** DIYMA 2021-08-19 · DIYMA 2021-08-21 · DIYMA 2021-12-02

In plain English: Many people think bass that seems to come from in front of you (not from the trunk) needs a big midbass playing very low. He says it depends on phase between the sub and midbass at their crossover. Phase describes where each speaker is in its push-pull motion; in phase means pushing together. A wrong match between the high-pass and low-pass is the number-one reason bass won't come from the front, and delay can't fix it. One trick (delay by half a wavelength, then flip polarity) only works at a single frequency. Aim to stay far from 180° (fully opposite) rather than hit exactly 0°: even 60° apart still adds up louder. Phase is hard to see directly, so check it by proxy: the measured slopes and the −6 dB points match the target, and the delays are correct. Separately, he says "impact" comes from around the midrange-to-tweeter crossover (about 3 kHz), not 40 Hz; in a 2021 post his examples are a bass guitar's thumb slap and the snap of toms.

3.7 #11 Door midbass crossed at 40 Hz is a red flag.

> Wishful-thinking crossovers (fronts at 40 Hz, sub at 60–70 Hz) were the biggest flaw in championship cars he helped. His six-woofer Harman experiment gave smooth bass in every seat but "no image in any seat," so he minimizes sub/midbass overlap. A high-passed midbass needs no box: the filter does the box's job. RARE
>
> **Sources:** DIYMA 2018-02-09 · DIYMA 2009 · DIYMA 2021-09-11

In plain English: He treats door midbass speakers set to play down to 40 Hz as a warning sign. He calls such settings wishful thinking: fronts down to 40 Hz with the sub crossed at 60–70 Hz, so both play the same bass. That overlap was the biggest flaw in championship competition cars he helped with. At Harman he tried six woofers: bass was smooth in every seat, but there was "no image in any seat" (the image is the sense that instruments sit at particular spots in front of you). So he keeps sub and midbass overlap small. He adds that a midbass with a high-pass filter needs

no box (enclosure): the filter does the box's job. Background: a sealed box acts on a speaker much like a high-pass filter, reducing its lowest notes and how far the cone moves; an electronic high-pass filter does that job instead.

3.7 #12 A low-pass on the sub alone is not a crossover.

> With full-range mains, the sub drifts out of phase with them about an octave above a 12 dB/oct cutoff (half an octave for 24 dB/oct). Crossovers work because the high-pass leads and the low-pass lags in a complementary way. RARE
>
> **Sources:** DIYMA 2022-02-10

In plain English: Some systems leave the main speakers full range (no high-pass) and only put a low-pass on the sub. He says that is not a real crossover. Every filter shifts timing (phase). In a true crossover the high-pass shifts its speaker ahead ("leads") and the low-pass shifts its speaker behind ("lags"), and the two shifts complement each other so the speakers add up correctly. With only a low-pass, nothing balances it: the sub drifts out of phase with the mains about an octave above a 12 dB/oct cutoff, or half an octave above with 24 dB/oct.

3.7 #13 "The real crossover is the acoustic output of our speaker."

> What counts is the measured point, slope, Q and phase (driver response × filter), not the DSP setting. Either EQ the driver flat first, or filter and then EQ until the result matches the target slope; then set delay by distance. RARE
>
> **Sources:** Audiofrog blog Jan 2021 (crossovers) · DIYMA 2017-06-25 · DIYMA 2020-10-19

In plain English: The number you type into the DSP is the electrical filter. What you actually hear is that filter combined with the speaker's own natural response in the car. He says the second is what counts: the measured crossover point, slope, Q (how sharp or gentle the bend is) and phase. There are two ways to get there: EQ the speaker flat first and then apply the filter, or apply the filter and then EQ until the measured result matches the target slope. Then set the delay by distance.

3.7 #14 The crossover point is where the measured curves cross, and channel level moves it.

> "Overlap" and "underlap" mean nothing unless measured; raising the low-passed channel's gain shifts the crossing (his example: sections meeting at 73 Hz with the dial at 63 Hz). Re-EQ after any crossover change, since crossovers are for safety and dispersion and EQ is for response. RARE
>
> **Sources:** DIYMA 2014 · DIYMA 2014 (73 Hz example) · DIYMA 2011

In plain English: He says the true crossover point is where the two speakers' measured curves cross. Words like "overlap" (both speakers play the same range) and "underlap" (a gap between them) mean nothing unless you measure. Turning up the gain (level) of the low-passed channel moves the crossing point. His example: the dial said 63 Hz, but the measured sections met at 73 Hz. So redo the EQ after any crossover change. His reasoning: crossovers are for safety and dispersion (how widely sound spreads); EQ is for the frequency response (the tonal balance).

3.7 #15 How filter types sum.

> Each 6 dB/oct adds 45° of shift. First order: 90° apart, flat on paper, poor in practice. 2nd-order Butterworth: +3 dB hump with polarity reversed (deep dip without); LR2: flat with reversal (180°). 3rd-order Butterworth: 270° apart, each −3 dB, flat in either polarity, so no polarity check. LR4: each −6 dB, 360° apart, flat with correct polarity.
>
> **Sources:** Audiofrog blog Jan 2021 (crossovers) · DIYMA 2018-01-21 · DIYMA 2022-01-14

In plain English: This point explains how different crossover recipes add together. Background: the "order" of a filter is its steepness: 1st order is 6 dB/oct, 2nd is 12, 3rd is 18, 4th is 24. Butterworth and Linkwitz-Riley (LR) are two families of filter shape. Polarity is which way the cone moves for a positive signal; swapping a speaker's + and − wires reverses it. He says each 6 dB/oct adds 45° of phase shift. First order: the two speakers are 90° apart; flat on paper, poor in practice. 2nd-order Butterworth: a +3 dB hump at the crossover with one speaker's polarity reversed, and a deep dip without the reversal. LR2: flat with the reversal, since the two are 180° apart. 3rd-order Butterworth: 270° apart, each 3 dB down at the crossover, flat in either polarity, so it gives you no way to check polarity. LR4: each 6 dB down, 360° apart (a full turn, so back in step), and flat with correct polarity.

3.7 #16 Steeper slopes don't protect more, and are no excuse to cross lower.

> Each 6 dB removes 75% of what's left (from 100 W the steps remove 75, 18.75, 4.69, then 1.17 W), so most protection comes early; a 48 dB/oct filter passes about 1.5× more power near the knee than LR4. Moving a tweeter from 4 to 2 kHz roughly doubles its power. 6 dB slopes are cheaper, not better; beyond 24 dB/oct wastes processing. RARE
>
> **Sources:** Audiofrog blog Oct 2021 (slopes and protection) · DIYMA 2021-07-15 · DIYMA 2013

In plain English: Many people believe a steeper filter protects a speaker better, so they can cross it lower. He says no. Each 6 dB of reduction removes 75% of the power that's left. From 100 W (watts), the first 6 dB step removes 75 W, the next 18.75 W, then 4.69 W, then 1.17 W, so most of the protection comes early. Near the "knee" (the corner where the slope begins), a 48 dB/oct filter actually passes about 1.5 times more power than LR4. The crossover point matters more: moving a tweeter from 4 kHz down to 2 kHz roughly doubles the power it receives. He says 6 dB/oct slopes are cheaper, not better, and going beyond 24 dB/oct wastes processing power.

3.7 #17 Tweeters see very little power.

> With 1000 W of pink noise in the 20 Hz octave, a mid crossed at 80 Hz gets at most ~250 W and a tweeter crossed at 2.5 kHz ~8 W; EDM sends even less up top, jazz more. A 24 dB/oct high-pass at 2.5 kHz is −24 dB at 1.25 kHz, so an extra series cap only acts where the tweeter contributes nothing. RARE
>
> **Sources:** Audiofrog blog Dec 2020 (pink noise) · DIYMA 2018-05-04

In plain English: Pink noise is a test signal with equal energy in every octave. He uses it as a stand-in for music and explains that, as a result, the power an amp delivers halves with each octave up: 1000 W at 20 Hz means 500 W at 40 Hz and 250 W at 80 Hz. So a midrange crossed at 80 Hz gets at most about 250 W, and halving all the way up to 2.5 kHz leaves a tweeter about 8 W. EDM

(electronic dance music) sends even less to the top; jazz sends more. Also, a 24 dB/oct high-pass at 2.5 kHz is already 24 dB down at 1.25 kHz, one octave lower. So an extra capacitor in series (a part that blocks low notes) only acts where the tweeter contributes nothing anyway.

3.7 #18 Keep a protection capacitor on tweeters even in active systems, but remove it if you go back to a passive crossover.

> Pick a value putting −3 dB about an octave below Fs (e.g., 68 µF for 4 Ω behind an LR4 2.5 kHz high-pass, ~0.5 dB change from 1.8–2.5 kHz); a ~$1.75 non-polar electrolytic is fine. It's a "parking brake" against pops, amp failures and wrong DSP entries; in a passive network it alters slope and Q. RARE
>
> **Sources:** Audiofrog blog Dec 2020 (to cap or not) · DIYMA 2021-09-22

In plain English: A capacitor wired in line with a tweeter blocks low notes. He says keep one on tweeters even in active systems (where the DSP does the crossover), but remove it if you go back to a passive crossover (the box of parts that often comes with component speakers). Choose a value that puts its −3 dB point about an octave below the tweeter's resonance (Fs). His example: 68 µF (microfarads) for a 4 Ω (ohm) tweeter behind an LR4 high-pass at 2.5 kHz, which changes the response by only about 0.5 dB between 1.8 and 2.5 kHz. A cheap non-polar electrolytic capacitor, about $1.75, is fine. He calls it a "parking brake" against pops, amplifier failures and wrong settings typed into the DSP. In a passive network, though, it changes the filter's slope and Q (the shape of its bend).

3.7 #19 Crossover summation checks only work anechoically; in a car, highs sum to about +3 dB.

> Reflections mean nothing is truly in phase up high, so L+R or driver sums won't reach +6 dB. Don't obsess over mid/tweeter phase (it can't be measured in a way that matches hearing); sub/midbass and midbass/mid phase matter and can be measured. RARE
>
> **Sources:** DIYMA 2021-03-26 · DIYMA 2021-08-25 · DIYMA 2021-07-12

In plain English: "Summation" means playing two sounds together (left plus right, or the two speakers either side of a crossover) and checking how much louder the result is. Background: two identical sounds perfectly in phase add up to +6 dB. He says this check only works anechoically, meaning in a space with no echoes. In a car, reflections mean nothing is truly in phase at high frequencies, so highs add up to only about +3 dB, and left-plus-right or driver sums won't reach +6 dB. His advice: don't obsess over midrange-to-tweeter phase, because it can't be measured in a way that matches what you hear. Phase at the sub/midbass and midbass/midrange crossovers does matter, and it can be measured.

3.7 #20 Group delay and FIR crossovers are overrated in cars.

> Stop-band group delay doesn't matter because the driver isn't used there; front-bass failures are crossover phase, not group delay (he mocks rejecting vented boxes for group delay, then using a 72 dB/oct low-pass at 40 Hz). FIR isn't inherently "frequency-dependent delay"; he questions FIR crossovers. RARE
>
> **Sources:** DIYMA 2017-09-12 · DIYMA 2022-01-14 · DIYMA 2021-08-20

In plain English: Group delay describes how much a filter or speaker delays some frequencies relative to others. FIR (Finite Impulse Response) is a type of digital filter often promoted for its phase behaviour. He says both are overrated in cars. Group delay in the stop-band (the range a

filter is blocking) doesn't matter, because the speaker isn't being used there. When bass won't come from the front, the cause is crossover phase, not group delay. He mocks people who reject vented (ported) boxes because of their group delay, then use a 72 dB/oct low-pass at 40 Hz, a very steep filter. He says FIR isn't inherently "frequency-dependent delay", and he questions FIR crossovers.

3.7 #21 Keep left and right crossover slopes identical.

> Different L/R midbass slopes act like an all-pass on one side, shifting relative phase about 45° an octave above the crossover and ~90° two octaves above. RARE
>
> **Sources:** DIYMA 2018-01-27

In plain English: Different crossover slopes on the left and right midbass act like an all-pass filter on one side. An all-pass filter leaves the level alone but shifts the timing (phase). The result: the left and right midbass drift about 45° apart one octave above the crossover, and about 90° apart two octaves above. So in practice: use the same slope on both sides.

3.7 #22 Sometimes you must move the crossover rather than EQ.

> Occasionally a reflection or cancellation forces it: in a Dodge Durango's OE Alpine system he moved door mids from ~4 kHz down to ~1 kHz, before they beamed and before their first big peak. Avoid bands under an octave wide. The popular "tricks" (big overlap, flipped mids, 6 dB slopes for "blend") always made his car worse.
>
> **Sources:** DIYMA MS-8 FAQ · DIYMA 2014 · DIYMA 2011

In plain English: Normally he fixes response problems with EQ, but occasionally a reflection or cancellation forces a crossover change instead. His example is a Dodge Durango with the factory (OE, original equipment) Alpine system. He moved the door midranges' crossover from about 4 kHz down to about 1 kHz, so they stopped before they began to beam and before their first big peak in response. He also says to avoid bands under an octave wide, meaning a speaker given a range of less than one octave to play; in a 2014 post he gives 80–150 Hz as an example of such a bad crossover. The popular "tricks" (big overlap between speakers, midranges with flipped polarity, and 6 dB slopes for "blend") always made his car worse.

3.7 #23 For a customer who only wants loud, raise the crossovers like a pro sound system.

> His 1970s-disco example: a "correct" 60–80 Hz system kept getting blown because the owner boosted bass ~12 dB at the head unit, past clipping, cancelling the midbass high-pass. Raise the crossover and accept less front bass; get to correct, then back off it. RARE
>
> **Sources:** CMA Connected (Oct 2021) @43:34

In plain English: Not every customer cares about imaging. His example is a customer who played 1970s disco at full volume. The dealer had built a "correct" system crossed at 60–80 Hz, but it kept getting blown. The owner boosted bass by about 12 dB at the head unit (the radio), pushing it past clipping (overloading so the signal's peaks get chopped off), and that boost cancelled the midbass high-pass. His fix: raise the crossover and set it up like a pro sound system, accepting less bass from the front. His wider point: "get to correct" first, then back off from correct to suit the listener.

3.7 #24 (MS-8 manual) Factory defaults, for reference.

> 24 dB/oct; sub/front 80 Hz; tweeters at 3.5 kHz or higher; subsonic 20 Hz 12 dB/oct; front high-pass without a sub 20–40 Hz for 6-in+ and 40–60 Hz for 5-in and smaller; center 80 Hz; sides and rears ≥100 Hz. Product documentation, not verified as his words. **Era:** MS-8
>
> **Sources:** MS-8 manual (mirror)

In plain English: This point is from the manual of the JBL MS-8, a car-audio processor made by Harman, where he worked on it. These are its factory starting settings, listed for reference: 24 dB/oct slopes; sub-to-front crossover at 80 Hz; tweeters at 3.5 kHz or higher; a subsonic filter (which cuts the very lowest notes) at 20 Hz, 12 dB/oct; front high-pass, when there is no sub, at 20–40 Hz for 6-inch speakers and larger and 40–60 Hz for 5-inch and smaller; center speaker 80 Hz; side and rear speakers 100 Hz or higher. As the point says, this is product documentation, not confirmed as his own words.

3.8 Polarity and phase

What this part is about: Polarity (which way a speaker's cone moves when the signal pushes) and phase (whether two speakers push and pull in step at a given pitch). It covers how he checks both, what phase targets he aims for, and how all-pass filters, including ones hidden inside factory systems, change the picture.

3.8 #1 Set polarity with markings or a test tool, never "whichever sounds better."

> Positive voltage should push every cone toward the listener. If a flip sounds better, find out why: a left/right distance error needs delay, not a flip. With LR4 (Linkwitz-Riley 24 dB/oct) and correct polarity the crossover sums flat, so a deep, narrow dip there flags a reversed driver; flip it on both sides. **RARE**
>
> **Sources:** Tuning PDF v2 (~2023) · DIYMA 2014 · DIYMA 2014

In plain English: Polarity is which way a speaker's cone moves when the signal goes positive; swapping a speaker's + and − wires reverses it. He says the right setting is that positive voltage pushes every cone toward the listener, and you set it from the wire markings or with a test tool, never by picking whichever sounds better. If a flip sounds better, find out why. For example, if the left and right speakers sit at different distances from you, the fix is delay (holding back the nearer speaker), not a flip. With LR4 (Linkwitz-Riley 24 dB/oct) and correct polarity, the crossover adds up flat. So a deep, narrow dip right at the crossover tells you a speaker is reversed; flip it on both sides.

3.8 #2 Never fix a vocal-range center problem by flipping one midrange, and never flip one and add delay too.

> His worked example: a flip "fixed" 250 Hz but worsened the bass sum; adding correct delays then left the midbasses 180° out at all frequencies and destroyed the center. Use delay or a flip, "one or the other, not both"; flipping is also what forum advice often tells people to do. **RARE**
>
> **Sources:** Tuning PDF v2 (~2023) · Audiofrog blog Dec 2020, Time Alignment Part 2 · OFCAP Ep. 19 @3:17:02

In plain English: The centre image is the sense that a singer stands in the middle, between the speakers. When it's wrong in the vocal range, forum advice often says to flip one midrange's polarity. He says never do that, and never flip one and add delay as well. In his worked example, a flip "fixed" 250 Hz but made the bass sum worse. Adding correct delays on top then left the midbasses 180° out of phase (fully opposed) at all frequencies, which destroyed the centre. His rule: use delay or a flip, "one or the other, not both".

3.8 #3 Most polarity-test tracks can't check woofers or subs; use a long, all-positive pulse.

> Short clicks lack the bass that passes a low-pass filter ("If there's no bass, then we can't check a subwoofer"). Use a right-triangle pulse with a vertical leading edge (Audiofrog's is about 9 ms) or a long train of positive pulses; read only the first motion (later negative swings are normal) and mute the other speakers. Alternative: a DC-offset track and a voltmeter at the speaker leads. RARE
>
> **Sources:** Audiofrog blog 2021-08-17, polarity · DIYMA 2021-08-16 · DIYMA 2015

In plain English: A polarity test track plays a pulse, and a polarity checker (a small tool held near the speaker) reads which way each cone first moves. He says most such tracks can't check woofers or subs, because their short clicks have little bass to pass through a low-pass filter: "If there's no bass, then we can't check a subwoofer." Instead, use a long pulse that is all positive: a right-triangle shape with a vertical leading edge (Audiofrog's lasts about 9 ms, nine thousandths of a second), or a long train of positive pulses. Read only the first motion, because later negative swings are normal, and mute the other speakers while you test. Another option: a DC-offset track (a steady positive voltage) and a voltmeter on the speaker's wires.

3.8 #4 A reversed tweeter in a passive crossover is usually deliberate.

> Only Linkwitz-Riley orders that are multiples of 4 sum in phase; 2nd order is 180° apart and needs a reversal. For a passive tweeter right next to the woofer (1.5" offset ≈ half a wavelength at 4.5 kHz), a deep narrow dip at the crossover means reverse the tweeter, then tune.
>
> **Sources:** Audiofrog blog 2021-08-17, polarity · DIYMA 2018-04-25 · DIYMA 2021-09-26

In plain English: A passive crossover is the set of parts (often in a small box) that comes with many component speaker sets. In these, the tweeter is often wired reversed, and he says that is usually deliberate. Only Linkwitz-Riley filters whose order is a multiple of 4 add up in phase; a 2nd-order filter leaves the two speakers 180° apart, so one needs reversing. For a passive tweeter right next to the woofer, he notes that even a 1.5-inch (about 38 mm) offset is about half a wavelength at 4.5 kHz (at that pitch, the two sounds arrive half a cycle out of step). His test: if there's a deep, narrow dip at the crossover, reverse the tweeter, then tune.

3.8 #5 Polarity is not delay.

> A flip starts the sound at the same moment and multiplies phase by −1 at every frequency; a fixed delay adds phase that grows with frequency (there are 360° in one cycle and f cycles in one second, so the same delay is a different number of degrees at each frequency). So a flip can never substitute for distance correction. RARE
>
> **Sources:** DIYMA 2008 · DIYMA 2016

In plain English: Polarity and delay are different tools, and he says one cannot stand in for the other. A polarity flip doesn't make the sound start later: it starts at the same moment, but every push becomes a pull (phase is multiplied by −1 at every frequency). A fixed delay holds the sound back, and the phase shift it adds grows with frequency. Every cycle has 360°, and higher notes complete more cycles each second, so the same delay amounts to a different number of degrees at different pitches. So in practice: a flip can never substitute for correcting a speaker's distance.

3.8 #6 The goal is "far from 180°," not 0°.

> Phase only matters where two speakers play the same frequency within about 12 dB; under 90° apart is hard to hear, so treat ±45° as in phase and leave anything within ±90° alone. Sub and midbass need only be within ±90°; if one midbass is in phase with the sub and the other isn't, shift the sub about 90° (+3 dB, not a null). RARE
>
> **Sources:** DIYMA 2012 · DIYMA 2016 · DIYMA 2021-02-04

In plain English: Phase difference is measured in degrees: 0° means two speakers push and pull exactly together, and 180° means exactly opposite, so they cancel. He says aim to be "far from 180°", not to hit 0°. Phase only matters where two speakers play the same frequency within about 12 dB of each other. Differences under 90° are hard to hear, so he treats ±45° as in phase and leaves anything within ±90° alone. Sub and midbass only need to be within ±90°. If one midbass is in phase with the sub and the other isn't, shift the sub's phase about 90°; that gives roughly +3 dB rather than a null (a deep cancellation dip).

3.8 #7 Read phase from the left+right sum of mono pink noise.

> In phase sums about +6 dB over one side; ~90° apart +3 dB; ~120° equals one side; less than one side means near 180°. Real cancellation is 135–225°. Use mono (correlated) noise; he sees no use for uncorrelated noise here. Every peak or dip carries a phase shift, so where L and R peaks don't line up, they aren't in phase. RARE
>
> **Sources:** DIYMA 2021-08-26 · DIYMA 2021-11-02 · DIYMA 2021-05-09

In plain English: This is his way to read phase from simple level measurements. Play mono pink noise (the same noise in both channels, called "correlated") and measure left alone, right alone, then both together. If the sum is about 6 dB louder than one side, they're in phase. About +3 dB means roughly 90° apart. If the sum equals one side, they're about 120° apart. If it's quieter than one side, they're near 180°. Real cancellation only happens between 135° and 225°. He sees no use here for uncorrelated noise (different noise in each channel). Also, every peak or dip in a response carries a phase shift, so where the left and right peaks don't line up, the two sides aren't in phase.

3.8 #8 In a car, highs only sum to about +3 dB, so the "summation" crossover method only works anechoically.

> Dense reflections average high-frequency phase out to about 90°; in 30 years he has never seen a car differ. Expect ~+6 dB below about 1 kHz and ~+3 dB above.
>
> **Sources:** DIYMA 2021-03-26 · DIYMA 2021-08-25 · DIYMA 2021-08-26

In plain English: Two identical sounds perfectly in step add up to +6 dB. In a car, he says, the highs only add up to about +3 dB, because dense reflections average the high-frequency phase out to about 90°. In 30 years he has never seen a car behave differently. So expect about +6 dB below

roughly 1 kHz and about +3 dB above it. That's why the "summation" method of setting crossovers (adjusting until the two speakers add up to the full +6 dB) only works anechoically, meaning in a space with no echoes.

3.8 #9 Phase matters much less above about 1 kHz; don't chase mid-to-tweeter phase.

> Those dips are narrow, an inch of head or mic movement changes them, and a spatial average has no phase at all. Set mid/tweeter delay by tape measure, EQ both to the target slopes, and check a few spots around your head. Sub/midbass and midbass/mid are the crossovers where phase is both important and measurable. RARE
>
> **Sources:** Audiofrog blog Dec 2020, Time Alignment Part 2 · DIYMA 2021-07-12 · OFCAP Ep. 19 @2:56:00

In plain English: Above about 1 kHz, he says, phase matters much less, so don't chase phase between the midrange and tweeter. The dips you see there are narrow, and moving your head or the microphone by an inch (25 mm) changes them. A spatial average (several measurements around the head combined into one) has no phase information at all. His method instead: set the midrange/tweeter delay with a tape measure, EQ both speakers to the target slopes, and check a few spots around your head. The crossovers where phase is both important and measurable are sub/midbass and midbass/midrange.

3.8 #10 When a crossover pair is in phase, you hear both drivers at the higher-frequency one.

> So correct sub/midbass phase pulls the bass forward, and correct midbass/mid and mid/tweeter phase keeps vocals at tweeter height ("the cure for the rainbow is a good crossover"). That is also why club-style listeners who want impact need a DSP: impact is midbass-to-sub phase. RARE
>
> **Sources:** DIYMA 2021-12-02 · DIYMA 2017-08-17 · CMA Connected 2021 @46:57

In plain English: When the two speakers either side of a crossover are in phase, you hear them as one sound, located at the higher-frequency speaker. So correct sub/midbass phase pulls the bass forward to the front speakers. Correct midbass/midrange and midrange/tweeter phase keeps vocals at tweeter height. In his words, "the cure for the rainbow is a good crossover". The sources don't define "rainbow" exactly; it appears to mean different frequency ranges seeming to come from different places, instead of all from one point above the dash. He adds that this is why club-style listeners, who want impact rather than imaging, still need a DSP: impact is midbass-to-sub phase.

3.8 #11 Only the acoustic crossover counts, and you can't flatten magnitude first and fix phase later.

> The target is an acoustic LR4, flat in magnitude and phase; EQ and electrical filters add together, and an all-pass that corrects phase also changes the summed level. For minimum-phase behaviour, matching one channel's magnitude to the other also matches phase.
>
> **Sources:** DIYMA 2013 · DIYMA 2017-06-25 · DIYMA 2015

In plain English: The "acoustic crossover" is what the speakers actually produce in the car, as opposed to the filter settings typed into the DSP. His target is an acoustic LR4: flat in both magnitude (level) and phase. He warns that you can't make the level flat first and fix the phase later. EQ and electrical filters add together, and an all-pass filter used to correct phase also

changes the level of the combined sound. "Minimum-phase" is a technical term for behaviour where a speaker's phase is tied to its level response. Where that holds, he says, matching one channel's magnitude to the other's also matches its phase.

3.8 #12 What an all-pass filter is: phase rotation with flat level, not a frequency-dependent delay.

> A 1st-order all-pass rotates 0–180° (90° at its frequency); a 2nd-order 0–360° (180° at its frequency); Q sets how fast. Its phase curve is a low-pass filter's. Degrees per inch depend on frequency: 1° ≈ 0.48" at 80 Hz, 1.86" at 20 Hz, 0.04" at 1 kHz. An "AP" crossover setting on an amp just means no filter. RARE
>
> **Sources:** Dean & Fernando live @10:07 · DIYMA 2021-12-03 · DIYMA 2021-05-13

In plain English: An all-pass filter lets every frequency through at full level but rotates its phase (shifts its timing) around a chosen frequency. A 1st-order all-pass rotates phase from 0° to 180°, reaching 90° at its set frequency; a 2nd-order one rotates from 0° to 360°, reaching 180° at its frequency. Q sets how fast the rotation happens. He notes its phase curve is a low-pass filter's, and stresses that it is not a frequency-dependent delay. How many inches one degree equals depends on frequency: 1° is about 0.48 inch (12 mm) at 80 Hz, 1.86 inches (47 mm) at 20 Hz, and 0.04 inch (1 mm) at 1 kHz. And an "AP" crossover setting on an amplifier just means no filter at all.

3.8 #13 A DSP's "phase" slider (e.g., Helix) is an all-pass; its degrees just move the filter frequency.

> At the sub/midbass crossover he'd rather get the crossover right; if he uses one, it goes on the high side of the crossover, where its effect on the sum shrinks as that speaker rolls off. All IIR filters (high-, low-, all-pass) delay frequencies around and below their frequency, never above.
>
> **Sources:** DIYMA 2017-07-07 · DIYMA 2021-12-03 · DIYMA 2021-03-05

In plain English: Many DSPs, such as the Helix, have a "phase" slider marked in degrees. He says it is really an all-pass filter, and changing the degrees just moves the filter's frequency. At the sub/midbass crossover he would rather get the crossover itself right. If he does use one, he puts it on the "high side" of the crossover; his reason is that where it rotates the phase, that speaker is getting quieter, so its effect on the combined sound shrinks. He adds a general rule: all IIR filters (the standard kind of DSP filter: high-pass, low-pass and all-pass) delay frequencies around and below their set frequency, never above.

3.8 #14 The 250 Hz insight: an all-pass gives both front seats a center, which delay can't.

> With door speakers the path difference first puts L and R out of phase around 250 Hz (lower in big cars, higher in small), mid-vocal range; the error is equal and opposite for the two seats at the same frequency, so reversing one side's phase in that band helps both. Driver's seat: a real but larger center (he likens delay-tuned cars to a diorama of tiny army men on the dash); passenger: diffuse but spread. RARE
>
> **Sources:** Dean & Fernando live @21:00 · Dean & Fernando live @23:05 · DIYMA 2016

In plain English: This is one of his key ideas for cars with two front seats. With door speakers, each front seat is closer to one side. That difference in path length first puts left and right out of phase around 250 Hz (lower in big cars, higher in small ones), in the middle of the vocal range. The error is equal and opposite for the two seats at the same frequency, so reversing one side's phase

in that band, with an all-pass filter, helps both seats at once. Delay can't do that. The result in the driver's seat is a real but larger centre image; he likens delay-tuned cars to a diorama of tiny army men on the dash. In the passenger seat the sound is diffuse but spread out.

3.8 #15 How he applies all-pass "phase EQ" for two seats.

> Use it only between about 80 Hz and 800 Hz–1 kHz, never higher. For a 250–500 Hz L/R mismatch: a 2nd-order all-pass at 250 Hz, Q about 1; in his own car, one on the sub plus one half an octave higher on one midbass centered the bass for both seats. To shift L/R phase across a band, put a 1st-order all-pass on each side at spaced frequencies. In a true two-seat system, drop the L/R delays.
>
> RARE **Changed view:** 2010 he had never made an all-pass work for sub/midbass blending ("all about amplitude"); 2014–15 Harman's sub-band phase EQ made "no audible difference"; from 2016 on it is his preferred two-seat tool.
>
> **Sources:** DIYMA 2010 · DIYMA 2016 · Dean & Fernando live @29:05

In plain English: "Phase EQ" here means using all-pass filters to adjust phase in chosen bands. He uses it only between about 80 Hz and 800 Hz–1 kHz, never higher. For a left/right mismatch between 250 and 500 Hz: a 2nd-order all-pass at 250 Hz with a Q of about 1. In his own car, one all-pass on the sub plus one half an octave higher on one midbass centred the bass for both seats. To shift left/right phase across a band, put a 1st-order all-pass on each side at spaced-apart frequencies. In a true two-seat system, drop the left/right delays. The [Changed view] tag: in 2010 he had never made an all-pass work for sub/midbass blending, which he called "all about amplitude" (level); in 2014–15 Harman's sub-band phase EQ made "no audible difference"; from 2016 on it has been his preferred two-seat tool.

3.8 #16 Before using an all-pass on an L/R dip, check whether it matters.

> If band-limited pink noise sits centered and the pan tracks move evenly, leave a small dip alone (his case: 650–1800 Hz, ~120° apart) or EQ both fronts identically. A dip that sums to the level of one side is ~120°, not 180°; with one mic, re-measure around a ~7" circle, and only if it persists nudge the all-pass frequency. An all-pass can fix one band and break another, and one on the driver's side helps only the driver. RARE
>
> **Sources:** DIYMA 2021-05-07 · DIYMA 2021-08-27 · DIYMA 2022-03-24

In plain English: Before using an all-pass on a left/right dip, he first checks whether the dip matters. Play band-limited pink noise (noise covering just the range in question) and pan tracks (test tracks that move a sound from side to side). If the noise sits centred and the pans move evenly, leave a small dip alone (his case: 650–1800 Hz, about 120° apart), or EQ both fronts the same way. A dip where the sum equals one side alone means about 120°, not 180°. With one microphone, re-measure at several points around a circle of about 7 inches (18 cm); only if the dip persists, nudge the all-pass frequency a little. Note that an all-pass can fix one band and break another, and one on the driver's side helps only the driver.

3.8 #17 Two quick tests tell you whether a factory system uses all-pass filters or delay.

> (1) Listen to a centered vocal from both front seats: all-pass gives a broad center from the driver's seat and both sides audible from the passenger seat; delay collapses everything into the passenger door. (2) Feed mono pink noise and view L+R in the DSP's input analyzer: one big dip between 200 and 500 Hz means all-pass; a dip followed by evenly spaced dips (a comb) means delay. RARE
>
> **Sources:** DIYMA 2021-10-26

In plain English: Factory systems sometimes use all-pass filters to give both front seats a centre image, while others use delay. He gives two quick tests to tell which. First, listen to a centred vocal from both front seats. With all-pass, the driver hears a broad centre and the passenger can hear both sides. With delay, everything collapses into the passenger door; he compares it to a one-seat competition tune heard from the wrong side. Second, feed mono pink noise and look at the left-plus-right sum in your DSP's input analyzer. One big dip between 200 and 500 Hz means all-pass; a dip followed by evenly spaced dips (a "comb", like a comb's teeth) means delay.

3.8 #18 What factory all-pass layouts look like, and why a polarity checker won't see them.

> Typical: one channel reversed plus a 1st-order all-pass that brings the bass back into phase, leaving the vocal band near 250 Hz centered and out-of-phase highs for width; or a 2nd-order all-pass near 250 Hz on one side; on one plot he read two 1st-order all-passes near 250 and 500 Hz. Bose chains several per channel. A pulse checker reads an all-pass like a crossover, so it shows no reversal. RARE
>
> **Sources:** DIYMA 2021-05-16 · DIYMA 2021-05-23

In plain English: Here he describes what factory all-pass setups typically look like. One common layout: one channel's polarity reversed, plus a 1st-order all-pass that brings the bass back into phase. That leaves the vocal band around 250 Hz centred, while the highs stay out of phase, which makes the sound wider. Another: a 2nd-order all-pass near 250 Hz on one side. On one measurement plot he read two 1st-order all-passes near 250 and 500 Hz. Bose chains several per channel. A polarity checker reads an all-pass the same way it reads a crossover filter, so it shows no reversal.

3.8 #19 Measure factory outputs against the original pink noise, not channel against channel.

> Comparing L with R can't show where a filter sits (X+Y=10 has many answers; 1+Y=10 has one). In REW an all-pass is easy to spot on a flat response with the start time set to zero. Check for all-pass filters, EQ and bass limiting before blaming the factory D/A.
>
> **Sources:** DIYMA 2021-05-20 · DIYMA 2021-05-16 · DIYMA 2021-02-03

In plain English: When investigating a factory system, he says to measure each output against the original pink noise signal, not left against right. Comparing two outputs with each other can't show which one holds a filter. His analogy: X + Y = 10 has many answers, but 1 + Y = 10 has only one; knowing the original signal is like knowing X. In REW (Room EQ Wizard, a free measurement program), an all-pass is easy to spot on a flat response with the start time set to zero. And before blaming the factory D/A (the converter that turns digital audio into an electrical signal), check for all-pass filters, EQ and bass limiting.

3.8 #20 Living with factory all-pass filters: keep the factory polarity and add no delays.

> "Reversed" OEM polarity is usually intended (connectors can't be swapped). An all-pass can't be inverted (that would break causality), and delay on top makes it work poorly; you can only match it with the same all-pass on the other side, or remove it with an FIR (Finite Impulse Response) filter that delays everything, adding latency that hurts Bluetooth calls. The driver still gets a broad image, and the passenger is spared a driver's-seat delay tune (see 3.13 #10). RARE
>
> **Sources:** DIYMA 2021-05-20 · DIYMA 2021-05-13 · DIYMA 2021-03-26

In plain English: If a factory system uses all-pass filters, he says keep the factory polarity and add no delays. "Reversed" factory polarity is usually intended: the factory connectors can't be swapped, so it isn't an installer mistake. An all-pass can't be inverted (undone by an opposite filter), because that would break causality: the filter would have to respond before the sound arrives. Adding delay on top makes it work poorly. Your only options are to match it with the same all-pass on the other side, or remove it with an FIR (Finite Impulse Response) filter, which delays everything and adds latency (lag) that hurts Bluetooth calls. The upside of leaving it: the driver still gets a broad image, and the passenger is spared a tune made with delays for the driver's seat (see 3.13 #10).

3.8 #21 "Pressure on the ears" usually means a low-frequency phase problem

> , especially in the driver's seat.
>
> **Sources:** DIYMA 2016

In plain English: Some listeners describe certain systems as putting "pressure on the ears". He says that usually means a phase problem in the bass range, especially in the driver's seat.

3.8 #22 Polarity markings on subs are almost never wrong.

> Claims that half are mismarked are implausible on a documented line; of ~4,000 subs built with reversible coloured binding posts, he knows of 3 wrong. With amp crossovers, make sure all are engaged at the same frequency.
>
> **Sources:** DIYMA 2018-04-07

In plain English: Polarity markings are the + and − labels or colours on a speaker's terminals. Some people claim half of subwoofers are mismarked. He says that's implausible for a documented product line: of about 4,000 subs built with reversible coloured binding posts (terminals whose colours could be fitted either way round), he knows of 3 that were wrong. If you use the amplifier's own crossovers, make sure they are all switched on at the same frequency.

3.8 #23 Different left and right slopes act like an all-pass on one side.

> Mismatched L/R midbass slopes shift relative phase about 45° an octave above the crossover and about 90° two octaves above. Alignment targets also differ by filter type: 0° at the crossover is right only for LR4; 1st-order should be 90° apart, LR2 −180°, and a 2nd-order Butterworth aligned at 0° peaks. RARE
>
> **Sources:** DIYMA 2018-01-27 · DIYMA 2018-01-21

In plain English: If the left and right midbass use different crossover slopes, one side behaves as if it had an all-pass filter (one that shifts timing without changing level). Their relative phase moves about 45° one octave above the crossover and about 90° two octaves above. He also says the right phase target depends on the filter type: 0° at the crossover is correct only for LR4. For 1st order the two speakers should be 90° apart, for LR2 −180°, and a 2nd-order Butterworth aligned at 0° gives a peak.

3.8 #24 Hearing midbass and tweeter as two separate sources means the crossover phase is wrong.

> **Sources:** DIYMA 2020-12-23

In plain English: If you can hear the midbass and the tweeter as two separate places the sound comes from, rather than one blended source, he says the crossover phase between them is wrong.

3.9 Time alignment and delay

What this part is about: Time alignment means adding tiny electronic delays so that sound from speakers at different distances reaches your ears at the same moment. This part covers what delay can and cannot do, how he measures and enters it, and the many situations where he says not to use it.

3.9 #1 "Delay doesn't move speakers."

> It only changes what happens between them; it can't move the left/right edges or push the stage forward or back (a friend leaving 30 minutes later doesn't move their house). Test: play the left channel alone and add delay; the image stays put. A DSP can only add delay, so you delay the nearer speakers to match the farthest. RARE
>
> **Sources:** Audiofrog blog Dec 2020, Time Alignment Part 1 · DIYMA 2021-03-26

In plain English: In a car you sit much closer to the speakers on your side, so their sound reaches you first. A DSP (digital signal processor, the small computer that shapes the sound before the amplifiers) can hold a speaker's signal back by a tiny amount; this is called delay or time alignment. Many people believe delay "moves" a speaker, pushing it farther away or pushing the stage forward. His point is that it doesn't. Delay only changes how the sound of two speakers combines in the space between them, for example where a centered voice appears. The outer edges of the stage stay at the speakers. His analogy: a friend who leaves home 30 minutes later hasn't moved their house. You can test it yourself: play only the left speaker and add delay; the sound stays exactly where it was. And because a DSP can only hold sound back, never speed it up, you delay the nearer speakers to match the farthest one.

3.9 #2 What delay is for: crossovers and the phantom center both assume equal arrival.

> A 27–28" left/right path difference (typical door mids) puts the first cancellation near 250 Hz, mid-vocal, repeating at odd multiples; delaying the near side about 2 ms (27 ÷ 13,582 in/s) fixes it at that seat. Delay also puts sub and fronts in phase: "nobody hates coherent bass."
>
> **Sources:** Audiofrog blog Dec 2020, Time Alignment Part 2 · Tuning PDF v2 (~2023) · OFCAP Ep. 38 @1:00:03

In plain English: Two things in a sound system only work if the sounds arrive together. A crossover splits the music between speakers (bass to the woofer, treble to the tweeter) and counts on the two halves arriving in step so they blend. The "phantom center" is the illusion of a singer floating between the left and right speakers, and it needs equal arrival too. With speakers in the doors, the far one is typically 27–28 inches (about 70 cm) farther from you. That gap means that near 250 Hz, the middle of a singing voice, one speaker's wave is pushing while the other's is pulling, so they cancel; the same cancellation repeats at 3, 5, 7 times that frequency. Sound travels about 13,582 inches per second, so 27 inches takes about 2 ms (two thousandths of a second). Delaying the near side by that much fixes it for that seat. Delay also lines up the subwoofer and front speakers so the bass works together: in his words, "nobody hates coherent bass."

3.9 #3 How to measure and enter delays.

> Put tape on the headrest center (or the mic tip) and measure to the center of each grille or dust cap. Key the front delays to the midrange, never an average of mid and tweeter ("just makes both wrong"). Subtract each distance from the largest (sub 70", tweeter 32" → tweeter gets 38", sub 0); ms = inches ÷ 13,584 × 1000. Know whether your DSP wants distance, time, or the subtracted value.
>
> **Sources:** Tuning PDF v1 (~2018) · DIYMA 2022-02-22 · DIYMA 2021-09-26

In plain English: This is his measuring procedure, step by step. First, put a piece of tape at the center of the headrest (or use the microphone tip) as the reference point for your head. Second, measure with a tape measure from there to the center of each speaker's grille or dust cap (the dome in the middle of a cone). Third, for the front speakers use the midrange's distance, not the tweeter's and not an average of the two; he says averaging "just makes both wrong." In the source post he gives his reasons: we hear phase errors more at lower frequencies, and the mids' delays help place the center image. Fourth, subtract each distance from the largest one. In his example the sub is 70 inches away and the tweeter 32 inches, so the tweeter gets 38 inches of delay and the sub, being farthest, gets none. Fifth, to turn inches into milliseconds, divide by 13,584 (the speed of sound in inches per second) and multiply by 1000; 38 inches is about 2.8 ms. Finally, check your DSP's manual: some want the raw distance, some a time, and some the subtracted value. Typing in the wrong kind is a common mistake.

3.9 #4 A tape measure is accurate enough; extra precision is wasted.

> At 80 Hz 1" ≈ 2° and 10" ≈ 20°; at 60 Hz a foot of error is only ~19°. A 1" error caused nothing in his midbass example, and LR4 forgives listeners of different heights. DSP resolution (one sample at 48 kHz ≈ 0.28") puts its cancellation at 24 kHz, so 1 mm resolution is "nonsense." Set delays before crossovers, then leave them alone. RARE
>
> **Sources:** Tuning PDF v1 (~2018) · DIYMA 2021-03-26 · DIYMA 2021-09-02

In plain English: Many people believe time alignment must be set to the millimeter, using a microphone and software. His point is that a tape measure is close enough. Background: "phase" describes where a sound wave is in its up-and-down cycle, measured in degrees; 360° is one full cycle, and 180° means exactly opposite, which cancels. Bass waves are long, so a small distance error barely shifts them: at 80 Hz an inch is about 2° and ten inches about 20°; at 60 Hz a whole foot of error is only about 19°. (He treats differences under 90° as hard to hear; see 3.8 #6.) In his midbass example a 1-inch error caused nothing, and an LR4 crossover (Linkwitz-Riley, 24 dB per octave, the type he recommends) forgives listeners of different heights. A DSP's smallest delay step at a 48 kHz sample rate is one sample, about 0.28 inch, and an error that size would only cancel at 24 kHz, above the range of human hearing, so 1 mm settings are "nonsense." So in practice: set delays before crossovers, then leave them alone.

3.9 #5 Impulse peaks aren't arrival times for low-passed drivers, so impulse alignment rarely beats a tape.

> The peak's slope depends on high-frequency content a sub or midbass doesn't play (REW's own help says so); the true arrival is the zero crossing before the right peak. If you must: crossovers off (midbass low-pass off), back up from the first tall peak to −12 dB, take the initial not the tallest peak. A $100 DSSF3 cross-correlation method costs hours and lands no closer. RARE
>
> **Sources:** DIYMA 2013 · DIYMA 2021-03-26 · OFCAP Ep. 19 @3:25:01

In plain English: An "impulse response" is a measurement of what the mic picks up after a speaker plays a very short click. Software such as REW (Room EQ Wizard, a free measurement program) draws it as a spiky graph, and many people read the tallest spike as the moment the sound arrived. His point is that for speakers that play only low notes (a sub, or a midbass with a low-pass filter that removes its highs), the peak is not the arrival time. How steep the peak is depends on high frequencies those speakers don't play; REW's own help says so. The true arrival is where the trace crosses zero just before the correct peak. If you must use impulses: switch the crossovers off (including the midbass low-pass), find the first tall peak, back up from it to the point 12 dB lower, and use the first peak, not the tallest. He says a cross-correlation method using DSSF3 (a $100 measurement program) costs hours and lands no closer than a tape measure.

3.9 #6 Automatic time alignment mis-measures dash speakers.

> It picks impulse peaks shaped by the highs, and a dash speaker's highs reach the mic via the windshield, so it measures speaker → glass → mic. Equal L/R errors are harmless; unequal ones cancel where the error is half a wavelength (a 2 kHz dip ≈ 3.5" ≈ 0.26 ms). RARE
>
> **Sources:** DIYMA 2022-03-24

In plain English: Some DSPs and measurement programs set delays automatically from a mic measurement. They look for peaks in the impulse response, and those peaks are shaped mostly by the high frequencies. A speaker on top of the dash usually fires up at the windshield, so its highs reach the mic mainly by bouncing off the glass. The software then measures the longer path, speaker to glass to mic, as if it were the speaker's distance. If the error is the same on the left and the right, no harm is done. If it differs, the two sides cancel at the frequency where the difference equals half a wavelength (half the length of one sound wave). His example: a dip at 2 kHz matches an error of about 3.5 inches (9 cm), or 0.26 ms.

3.9 #7 The invert-and-find-the-null test rarely works.

> It needs even-order crossovers, closely matched levels and both drivers EQ'd to the target slopes an octave each side, and reflections can still spoil it; it suits 12 dB Butterworth, not LR4.
>
> **Sources:** DIYMA 2021-09-02 · DIYMA 2015

In plain English: A popular trick for setting delay between two speakers that share a crossover works like this: reverse one speaker's polarity (its electrical plus and minus), play sound around the crossover frequency, adjust delay until the sound is quietest (the "null"), then switch the polarity back. He says this rarely works. It needs even-order crossovers (filter slopes such as 12 or 24 dB per octave), closely matched levels, and both speakers already equalized to the intended filter shape for an octave (a doubling or halving of frequency) on each side of the crossover. Even then, reflections in the car can spoil the result. It suits a 12 dB per octave Butterworth crossover, not the LR4 type he recommends. In the source thread his advice is to compensate for distance with a tape measure instead.

3.9 #8 Never set delays by ear or use them to move the image or add "depth."

> Ear-setting needs matched level and response that usually don't exist yet; delay tweaks move the image but "screw up" other frequencies. He rejects score-sheet advice to "play with delays" for stage

> depth. Of TrackRite-style tools, only the distance part works. RARE
>
> **Sources:** Tuning PDF v2 (~2023) · DIYMA 2018-05-22 · OFCAP Ep. 19 @3:23:07

In plain English: Many people believe the way to tune is to sit in the car and nudge delays by ear until the singer sounds right, or to use delays to make the stage seem deeper. He says never. Setting delay by ear only works if left and right are already matched in level and frequency response, and at that stage they usually aren't. Changing a delay can move the image, but it "screws up" other frequencies. He rejects the advice on competition score sheets (the judges' forms) to "play with delays" for stage depth. TrackRite is a commercial tuning aid; of tools like it, he says only the part that sets delays from distance works, not the parts meant for crossovers or for moving the image.

3.9 #9 Delay is for distance only, never for crossover phase or group delay.

> Music is mostly steady state, so there is no need to make up the cycle of lag across a 24 dB crossover. Car left and right speakers differ, so re-tweaking mid delays for crossover phase likely ruins the center ("a haircut using a cartoon image"). Use distance delays, LR4 and EQ each driver to that acoustic target. RARE
>
> **Sources:** DIYMA 2015 · DIYMA 2021-03-26

In plain English: Background: a crossover filter doesn't only divide frequencies; it also makes some of them lag slightly behind others. This lag is called group delay, and across a 24 dB per octave crossover it adds up to about one full cycle. Some tuners add extra delay to make up for it. He says delay is for distance only. Music is mostly "steady state" (sustained sound rather than isolated clicks), so there's no need to make up that lag. And because a car's left and right speakers sit in different spots and behave differently, re-tweaking the midrange delays to fix crossover phase will likely ruin the center image. His full comparison is giving yourself a haircut using a cartoon of you drawn by someone else: you are working from a picture that doesn't match reality. So in practice: set delays by distance, use LR4 crossovers, and EQ each speaker until its measured response matches that LR4 shape.

3.9 #10 Reflections, cabin gain, corner loading and resonances don't change delay settings.

> As long as a direct path exists, the direct sound arrives first and sets the delay; a reflection matters only if it is louder than the direct sound (only possible for a tweeter aimed away). Reflections make EQ useful, not different delays. The one judgement call: a narrow spike followed by a bigger, wider one may be an adjacent reflection, so find the surface first. RARE
>
> **Sources:** DIYMA 2021-03-26 · DIYMA 2013

In plain English: Background: "cabin gain" is the bass boost a small closed space like a car gives; "corner loading" is the boost a speaker gets near surfaces; resonances are frequencies where parts of the car ring. Many people think these call for different delays. He says they don't. If a speaker has a clear straight line to your ears, its direct sound always arrives first, and that sets the delay. A reflection (a bounce off glass or a panel) only matters if it is louder than the direct sound, which can happen only with a tweeter aimed away from you. Reflections are a reason to use EQ, not to change delays. The one judgement call: on an impulse graph, a narrow spike followed by a bigger,

wider one may be the direct sound plus a reflection from a nearby surface. In the source post he says to check with a tape measure and find that surface before deciding which spike is the real arrival.

3.9 #11 Align drivers within one side too; only their overlap band combs.

> At 1 kHz (13.6" wavelength) a mid 6.8" farther than the tweeter is 180° out; 0.5 ms on the tweeter fixes it. Once a cancellation is pushed above audibility or outside the pair's passband it doesn't matter, so small midbass errors below 200 Hz are inaudible. Aim for everything to seem to come from one point above the dash, no "rainbow."
>
> **Sources:** Audiofrog blog Dec 2020, Time Alignment Part 3 · DIYMA 2020-10-03

In plain English: Delay isn't only for left versus right. On the same side, a midrange and a tweeter at different distances can also interfere, but only in the band where both play (the overlap around their crossover). That interference is called comb filtering: a row of evenly spaced dips. His example: at 1 kHz one wave is 13.6 inches long, so if the midrange is 6.8 inches (half a wave) farther away than the tweeter, the two are 180° apart and cancel; 0.5 ms of delay on the tweeter fixes it. Once a cancellation is pushed above what we can hear, or outside the range the pair plays together, it no longer matters, so small errors between midbass speakers below 200 Hz are inaudible. His goal: everything should seem to come from one point above the dash, with no "rainbow." The sources don't define "rainbow" exactly; it appears to mean different frequency ranges seeming to come from different places.

3.9 #12 Don't time-align the fronts to the sub.

> Harman's repeated trials found no audible or measurable difference in bass placement below a 100 Hz crossover; an 80 Hz wave is ~14 ft, so a sub must be ~7 ft farther to be 180° off (a polarity swap fixes that). If bass seems behind you, check 60–160 Hz for dips, lower the sub, or cross sub 60 Hz/midbass 80 Hz, since raising sub level moves the acoustic crossover; last resort, delay all fronts equally. RARE **Changed view:** he argued "over and over" for sub alignment at Harman, then lost. **Conflict:** the Tuning PDFs tape-measure the sub like any other speaker.
>
> **Sources:** DIYMA 2013 · DIYMA 2021-03-26 · Audiofrog blog Dec 2020, Time Alignment Part 3

In plain English: Harman is the large audio company (owner of JBL and other brands) where he worked. A sub in the trunk is much farther away than the front speakers, so many people delay the fronts to line up with it. He says don't. Harman's repeated tests found no audible or measurable difference in where the bass seemed to come from, with the sub crossed below 100 Hz. Bass waves are long: an 80 Hz wave is about 14 feet (4.3 m), so the sub would need to be about 7 feet farther away to be fully opposite (180°), and a polarity swap would fix that. If the bass seems to come from behind you, he suggests checking 60–160 Hz for dips, turning the sub down, or crossing the sub at 60 Hz and the midbass at 80 Hz, because turning the sub up shifts where the two actually blend (the acoustic crossover). Last resort: delay all front speakers by the same amount. The tags note that he argued for sub alignment at Harman "over and over," then lost; and that his Tuning PDFs measure the sub with a tape like any other speaker, which conflicts with this advice.

3.9 #13 Factory delay and DSP latency both sneak into your timing.

> Most factory systems with outboard amps already contain time alignment, and almost no "un-EQ" integration DSP removes it; the leftover delay combs and makes your settings wrong. Run every amp through the processor: MS-8's ~8 ms latency means a sub fed straight from the head unit arrives early, and total latency can cause hands-free echo. RARE
>
> **Sources:** DIYMA 2013 · DIYMA MS-8 thread · DIYMA (MS-8 replacement)

In plain English: Factory systems with a separate amplifier often contain their own processing. "Integration" DSPs take the factory signal and try to "un-EQ" it, undoing the factory sound shaping so you start clean. He says most factory systems with outboard amps already include time alignment, and almost none of these DSPs remove it. The leftover factory delay causes comb filtering and makes your own settings wrong. Second, every processor takes a little time to work, called latency; the MS-8 (a JBL processor from his Harman years) takes about 8 ms. If one amp, such as the sub's, is fed straight from the head unit (the radio) while the others go through the processor, the sub's sound arrives early. So in practice: run every amp through the processor. Total latency can also cause an echo on hands-free phone calls.

3.9 #14 Delay is a one-seat tool: it doubles the other seat's error.

> With ~100 cm vs ~150 cm paths, delaying the near side ~1.5 ms doubles the passenger's error. For two seats use placement (kick panels or any spot that keeps the path difference under about a foot), all-pass phase EQ (3.8 #14–15) or a center channel. **Changed view:** "forget the passenger" is a cop-out; a good passenger seat no longer costs the driver's seat.
>
> **Sources:** DIYMA 2016 · DIYMA 2021-06-29 · OFCAP Ep. 38 @52:04

In plain English: Delay fixes the timing for one seat only. Say the driver is about 100 cm (39 inches) from the left speaker and 150 cm (59 inches) from the right. Delaying the left side by about 1.5 ms fixes the driver's seat. But the passenger already hears the right speaker first, and the delay makes the left one arrive even later for them, so their error doubles. For good sound in both seats he points to speaker placement (kick panels, the panels in the footwells, or any spot that keeps the left/right distance difference under about a foot), all-pass "phase EQ" (see 3.8 #14–15), or a center channel. The tag marks a change of view: he now calls "forget the passenger" a cop-out, because a good passenger seat no longer has to cost the driver's seat anything.

3.9 #15 With a center channel and upmixer, drop the left/right delays.

> The center carries the center image, and center-to-side path differences are small, pushing phase problems up where they matter less; EQ left, center and right alike and image with level only. Toole's no-delay-with-a-center advice works only if the center covers all the midbass the mains play. RARE
>
> **Sources:** SQOLOGY Part 2 @14:02 · DIYMA 2015

In plain English: A center channel is a speaker in the middle of the dash. An upmixer is processing (such as Logic 7 or Dolby Pro Logic II) that pulls centered sounds out of a stereo recording and sends them to that speaker. He says that with this setup you remove the left/right delays. The center speaker now carries the center image itself, so delay is no longer needed to create one. The distance differences between the center and each side speaker are small, which pushes any phase problems up to higher frequencies where they matter less. EQ the left, center and right speakers

to sound alike, and place the image with level only. Floyd Toole, a well-known loudspeaker researcher, advises no delay with a center; he says that works only if the center plays all the midbass the left and right speakers play.

3.9 #16 A wide ~6 dB dip in the L+R sum can be fixed by trimming the driver-side delay while you watch it.

> First make sure it isn't one mic spot (move the mic; check single readings). A 2nd-order all-pass suits narrow, deep dips; for a wide one, reduce the driver's-side delay with both sides playing and keep the change if nothing else breaks. RARE
>
> **Sources:** DIYMA 2022-03-24

In plain English: "L+R sum" means measuring the left and right sides playing together. A dip of about 6 dB (a clear drop in loudness) spread over a wide range of frequencies suggests the two sides are partly out of step there. First make sure it's real: if you measured at one mic spot, move the mic a little; if you used an average, take several single readings around the area. If the problem shows up in only one spot, ignore it. A 2nd-order all-pass filter (a phase-shifting filter, see 3.8 #12) suits narrow, deep dips. For a wide one, he says: with both sides playing, reduce the driver's-side delay while watching the dip on screen, and if it improves without creating another problem, keep the change.

3.9 #17 To "de-tune" for listeners who dislike a center image, move delays deliberately; never switch them off.

> Active 3-way: delay the midbass but not mids and highs. 2-way plus sub: shorten the near-side delay so drivers stay in phase near the ~100 Hz crossover but not above, or flip mids and tweeters while keeping midbass in phase. Save the correct tune as a second preset. Correct is the start of expertise; backing off it to please a listener is the rest. RARE
>
> **Sources:** OFCAP Ep. 38 @1:02:06 · CMA Connected 2021 @42:16

In plain English: Some listeners dislike a tightly centered image (see 3.10 #7). For them he deliberately loosens the tune instead of switching delays off; he says switching them off costs bass coherence. He explains that in phase you hear an image between the speakers, and out of phase you hear the speakers themselves. In an active 3-way (separate amp channels for midbass, midrange and tweeter): delay the midbass, but not the midrange and tweeter. In a 2-way plus sub: shorten the near-side delay so the speakers stay in step around the roughly 100 Hz crossover but not above it, or flip the polarity of the mids and tweeters while keeping the midbass in phase. Save the correct tune as a second preset so you can switch back. His view: getting it correct is the start of expertise; knowing how to back off from correct to please a listener is the rest.

3.9 #18 Rear fill: plain in-phase stereo, turned up until it interferes, then delayed 10–20 ms.

> High-pass the rears at ≥100–200 Hz; add delay until they stop sounding like separate sources (more level needs more delay; ~35 ms becomes an echo; 1 ms ≈ 1 ft); in small cars set level and delay per side. RARE **Changed view:** in 2017 he still gave recipes for differential rear fill (L−R, delayed 12–20 ms); by late 2021 he said he'd helped popularize it but now finds it nasal, phasey and hard to tune.
>
> **Sources:** DIYMA 2017-10-06 (old view) · DIYMA 2021-12-02 · OFCAP Ep. 19 @3:33:00

In plain English: "Rear fill" means playing some sound through the rear speakers for a sense of space while the image stays in front. His later (2021) recipe: feed the rears ordinary stereo, in phase. Tune the front first, then turn the rears up until they start to interfere with the front image, and stop. High-pass them (remove their bass) at 100–200 Hz or higher. Add delay until they no longer sound like separate sources and the image is back in front; 10–20 ms (hundredths of a second) is about right, and more level needs more delay. At around 35 ms the delay becomes a distinct echo. Sound travels roughly 1 foot per millisecond. In small cars, set level and delay for each side separately. The tag: in 2017 he still gave recipes for "differential" rear fill (the difference signal, left minus right, delayed 12–20 ms); by late 2021 he said he had helped popularize it but now finds it nasal, phasey and hard to tune.

3.9 #19 Measure an L−R (difference) feed one side at a time, and treat center leakage as a clue.

> Mono noise into L−R gives silence, so play one channel or pan the balance fully (L − 0 = L). Center content leaking into an L−R output means something upstream isn't matched: noise not truly mono, unequal input EQ or levels, head-unit phase or delay differences, or lossy noise files.
>
> **Sources:** DIYMA 2021-05-11 · DIYMA 2021-12-02

In plain English: An L−R (left minus right) signal contains only what differs between the two channels; anything identical in both cancels out. So if you measure such an output with mono pink noise (the same hiss-like test signal in both channels), you get silence. Instead, play just one channel, or turn the balance fully to one side: with nothing on the right, L minus 0 is simply L. If centered sound leaks into an L−R output, he says something earlier in the chain isn't matched: the noise isn't truly mono, the input EQ or levels differ between left and right, the head unit (radio) adds different phase or delay to each side, or the noise files are lossy (compressed, like an MP3).

3.9 #20 Stage "width" from a delay error is just out-of-phase combing.

> An extra 1 ms on one side starts cancellations at 500 Hz, then 1.5 kHz, 2.5 kHz and so on; in those bands you hear the two speakers, which passes for width, and in between they fuse into an image. RARE
>
> **Sources:** DIYMA 2017-10-01

In plain English: Some people like the "wide" stage they hear when delays are off or wrong. He says that width is just an error. If one side has an extra 1 ms (one thousandth of a second) of delay, the two sides cancel at 500 Hz, then again at 1.5 kHz, 2.5 kHz and so on, a pattern called comb filtering. In those bands you hear the two speakers as separate sources, which passes for width; in the bands between, they fuse into a single image. So what sounds like extra width is really this out-of-phase pattern.

3.9 #21 How judged cars game the score, and why that isn't the goal for customers.

> Competitors optimize per judging track; without per-track EQ he'd skip delay and use an all-pass for the center-vocal track. Everything above the vocal fundamental (about 100–250, maybe 300 Hz) can be out of phase without moving the vocal, and judges hear those out-of-phase highs as stage width. Winning isn't great audio; for customers, get close to correct. RARE
>
> **Sources:** OFCAP Ep. 38 @54:01

In plain English: In sound-quality competitions, judges score cars on specific test tracks, and competitors tune for each judging track. He says that if he couldn't switch EQ between tracks, he would skip delay and use an all-pass filter (a phase-shifting filter) for the centered-vocal track. A voice's "fundamental" is its lowest, main pitch, here about 100–250 Hz, maybe 300 Hz. Everything above it can be out of phase without moving the vocal from the center, and judges hear those out-of-phase highs, where the separate speakers become audible, as a wide stage. He adds that he has sat in cars scored for a "wide stage" that had no center image on ordinary music. His conclusion: winning isn't the same as great audio; for customers, get close to correct.

3.10 Level matching and centering the image

What this part is about: After the delays, the next job is making the left and right sides equally loud at every frequency, so that a centered singer really sounds centered. This part covers how he matches levels, how he moves the center image, and why a centered image is a goal not every listener shares.

3.10 #1 Delay is the easy half; matching left and right level and response is the hard half.

> Time alignment fixes only the phase half; in one member's car the channels differed by almost 20 dB at 80 Hz, which no delay can fix. Center-panned sound is mono, so each ear needs identical level from both sides at every frequency; "4 or 5 bands of parametric EQ aren't enough." Match overall L/R within "a couple of dB" with channel levels before EQ. RARE
>
> **Sources:** DIYMA 2008 · DIYMA 2017-09-12 · Tuning PDF v1 (~2018)

In plain English: A centered image needs two things: the left and right sounds must arrive in step (phase, which delay handles) and be equally loud. "Frequency response" is how loud a speaker plays at each pitch, from deep bass to high treble. He says delay is the easy half. The hard half is making left and right match in level and response. In one forum member's car the two sides differed by almost 20 dB at 80 Hz, a huge difference that no delay can fix. A singer panned to the center is mono, meaning the same signal in both channels, so each ear needs the same level from both sides at every frequency. That takes a lot of EQ: "4 or 5 bands of parametric EQ aren't enough." (A parametric EQ band is one adjustable boost or cut at a chosen frequency.) So in practice: first get the overall left/right levels within "a couple of dB" using the channel level controls, then EQ.

3.10 #2 Center the image with level, never delay: move the whole left or right front.

> If the center sits left, turn the entire left side down a little (and vice versa); ignore what the level controls end up reading. Off-center in a left-hand-drive car, expect the right side to need more level. Level, unlike delay, can't be set by measurement alone: the mic lumps direct sound and reflections together, while we hear the direct sound as location. Used last, it has worked in "hundreds of cars." RARE
>
> **Sources:** Tuning PDF v2 (~2023) · DIYMA 2022-03-19 · DIYMA 2022-03-01

In plain English: Once delays and EQ are done, if a centered singer still leans to one side, he moves it with level, never delay. If the center sits to the left, turn the whole left front side (all its speakers together) down a little; if it sits right, turn the right side down. Ignore what the level controls read afterward. In a left-hand-drive car, where you sit to the left, expect the right side to need more level. Some argue that needing to lower one side means the delays are wrong; he

disagrees. Delay can be set by measurement, but level can't: a microphone hears the direct sound and all its reflections as one sound, while our ears take the location from the direct sound. So a meter can read "equal" when your ears disagree. He says doing this last has worked in "hundreds of cars."

3.10 #3 Diagnose a missing or off-center image before touching levels.

> A smeared or one-sided singer usually means delays entered wrong (e.g., distances typed into a DSP that wants delays), a reversed mid or midbass, or a left/right response mismatch; recheck delays with the tape, then the L/R match, and only then level.
>
> **Sources:** Absolute Electronix livestream @43:32 · DIYMA 2017-10-31 · DIYMA 2018-05-22

In plain English: If the singer sounds smeared across the dash or stuck on one side, don't reach for the level controls first. He says the usual causes are: delays entered wrong (for example, typing measured distances into a DSP that expects delay values), a midrange or midbass wired with reversed polarity (plus and minus swapped), or left and right not matching in frequency response. His order: re-check the delays with the tape measure, then check the left/right match, and only then adjust level.

3.10 #4 To center on the driver instead of the dash, turn the far side down about 6 dB.

> Include the sub if it has two inputs; he says this moves the image about halfway between dash center and the left speaker. With arrival and level matched, the center lands midway between the speakers, not in front of the driver, and a 6 dB difference moves it halfway toward the louder side. A near speaker at 18" vs a far one at 36" is already about 6 dB louder, which feels like too much when set by hand. RARE
>
> **Sources:** Tuning PDF v1 (~2018) · DIYMA 2017-07-07 · DIYMA 2012

In plain English: With delays set and levels matched, a centered singer appears midway between the two speakers, in the middle of the dash, not straight in front of the driver. Some people want it in front of the steering wheel instead. He says to turn the far side down about 6 dB (a 6 dB cut is a quarter of the power), including the sub if it takes two inputs (left and right). A 6 dB difference moves the image halfway toward the louder side, so he says it lands about halfway between the dash center and the left speaker. Distance alone also changes level: a speaker 18 inches away is about 6 dB louder at your ear than one 36 inches away. In a 2012 post he says matching those takes about a 6 dB cut on the near side, which can feel like too much when you set it by hand.

3.10 #5 The phantom image is vector math: balance only steers where the speakers are in phase.

> Without delay correction the summed vector leans to the nearer speaker; with it, the simple level-based sum returns. Even with correct delay, a 6 dB L/R difference over half an octave pulls that band toward the louder side, and level-based localization works well into the midrange, not just highs. RARE
>
> **Sources:** DIYMA 2015 · DIYMA 2015 · DIYMA 2018-05-22

In plain English: A "vector" is a quantity with both a size and a direction, like an arrow. His point is that the phantom image comes from adding the left and right sounds like arrows, which takes both their level and their timing into account. With delays uncorrected, the combined result leans toward the nearer speaker. With delays corrected, the simple rule returns: the louder side pulls the

image toward it, which is what the balance control relies on. Even with correct delays, if one side is 6 dB louder over just half an octave (a narrow slice of frequencies), that slice is pulled toward the louder side. A common belief is that level differences only steer high frequencies; he says level-based localization works well into the midrange too.

3.10 #6 Last step, noise and gain: move every amp gain by the same amount.

> If hiss on the silent track bothers you, lower all amplifier input-sensitivity controls equally; for more volume raise them equally until hiss appears, then back off. In the attributed gain guide, amp gains are set acoustically against the target (DSP channels at −12 dB), not with a scope, tweeter amp down first; the sub is raised to target with mono noise, and if only one input feeds the sub expect to add about 6 dB.
>
> **Sources:** Tuning PDF v2 (~2023) · DIYMA gain guide (attributed)

In plain English: An amplifier's gain (input-sensitivity) control sets how strongly the amp responds to the incoming signal. It isn't a volume knob, but it affects how loud the system can play and how much background hiss you hear. His final step: play the silent track on the test disc. If the hiss bothers you, lower every amp's gain by the same amount, so the balance you tuned doesn't change. If you want more volume, raise them all equally until hiss appears, then back off. A forum guide titled "Andy's guide to setting gains" (credited to him only by its title) sets amp gains acoustically, by measuring the sound in the car against the target curve, not with an oscilloscope (an instrument that displays voltage). In it, the DSP channels sit at −12 dB and the tweeter amp is turned down first; the sub is raised to the target using mono noise, and if only one input feeds the sub, expect to add about 6 dB.

3.10 #7 A centered image is a preference, not a universal goal.

> Two famous record producers heard his demo car: one loved the imaging, the other heard the center as "mono" and hated it. People who prefer processing off "aren't used to hearing true stereo" ("audiophile mode" vs "amusement park mode"); give them a de-tuned preset (3.9 #17).
>
> **Sources:** CMA Connected 2021 @41:19 · DIYMA 2011

In plain English: Not everyone wants a sharp, centered singer. Two famous record producers listened to his demo car: one loved the imaging; the other heard the center as "mono" and hated it. He says people who prefer the processing switched off "aren't used to hearing true stereo." He joked about an "audiophile mode" and an "amusement park mode." So in practice: give such listeners a de-tuned preset, as described in 3.9 #17.

3.10 #8 Many cars image well quietly and poorly loud.

> Cabin reflections spread images, more so at high volume.
>
> **Sources:** DIYMA 2010 · DIYMA 2022-02-23

In plain English: Reflections are sound bouncing off the glass and panels before it reaches you. He says they spread images out, more so at high volume. So a car can image well at low volume and poorly when turned up.

3.10 #9 He tolerates frequency-response errors more than spatial ones.

> A peak or dip bothers him less than a missing center image.
>
> **Sources:** SQOLOGY Part 2 @46:07

In plain English: A frequency-response error is a tonal one: some pitches too loud (a peak) or too quiet (a dip). A spatial error is about where sounds appear. He says a peak or dip bothers him less than a missing center image, so he puts up with tonal errors more readily than spatial ones.

3.11 Listening checks and test tracks

What this part is about: Measuring sets up the tune; listening checks it. This part covers when he listens, what sounds he uses (band-limited pink noise, panned bass guitar and piano rather than sine tones or singers), and what a good result should and shouldn't be expected to do.

3.11 #1 Listening verifies the tune; it doesn't set it up.

> He never listens until the measurements say the car is ready, and doesn't iterate by ear between steps ("you don't taste raw chicken after salting it"). Early listening misleads because small level differences are hard to hear and delay or response errors shift the image. If hours of touch-up follow the objective steps, the measurement, the tuner's grasp of it, or the process is wrong.
>
> **Sources:** DIYMA 2017-06-28 · UMI-1 video @24:03 · DIYMA 2021-07-09

In plain English: Many tuners adjust a little, listen, and adjust again, over and over. He doesn't. He measures and sets everything first, and only listens once the measurements say the car is ready; listening is the check at the end, not the method. He doesn't go back and forth by ear between steps. His analogy: you don't taste raw chicken after salting it. Listening early misleads you, because small level differences are hard to hear, and delay or response errors shift the image around. And if a tune still needs hours of touch-up by ear after the measured steps, something is wrong: the measurement, the tuner's understanding of it, or the process itself.

3.11 #2 The first reference a tuner needs is the sound of two speakers in phase, not "a concert."

> Many tuners have never sat between two speakers and flipped one to hear the difference. Training method: buy any matched $5 thrift-store pair, set them in an equilateral triangle, sit dead center and play music; imaging is just what two similar speakers do for a centered listener. RARE
>
> **Sources:** Absolute Electronix livestream @41:56 · Absolute Electronix livestream @46:15

In plain English: Many people believe a tuner's goal is to make a car sound like a live concert. He says the first reference you need is simpler: the sound of two speakers in phase. Many tuners have never sat between two speakers and flipped one (reversed its polarity) to hear the difference. His training method: buy any matched pair of speakers, even a $5 pair from a thrift store, set them up so they and you form an equilateral triangle (all three sides equal), sit dead center and play music. That teaches you what imaging is: simply what two similar speakers do for a listener sitting in the middle.

3.11 #3 Check placement band by band with mono band-limited pink noise; each band should sit centered.

> In the 2021 video: track 2 (20–160 Hz) must come from the dash center in front of you, track 3 (80–400 Hz) checks midbass, then 400 Hz–1 kHz and 400 Hz–20 kHz. Highs spread more, which is acceptable (dash and windshield reflections). For a quick center check, mono pink noise band-passed 100 Hz–3 kHz. **Conflict:** the Tuning PDFs number these tracks 2–5.
>
> **Sources:** UMI-1 video @27:01 · Tuning PDF v2 (~2023) · DIYMA 2011

In plain English: Pink noise is a hiss-like test sound containing all frequencies. "Band-limited" means filtered so only one range plays, and "mono" means identical in both channels, so it should appear dead center. The tracks referred to are on the test disc that goes with his tuning process. In his 2021 video: track 2 (20–160 Hz, the bass) must come from the dash center in front of you; track 3 (80–400 Hz) checks the midbass; further tracks cover 400 Hz–1 kHz and 400 Hz–20 kHz. Each band should sit centered. Higher frequencies spread out more, which is acceptable because of reflections from the dash and windshield. For a quick center check, use mono pink noise band-passed (filtered) to 100 Hz–3 kHz. The tag flags that the Tuning PDFs number these tracks 2–5, differently from the video.

3.11 #4 Use panning tracks of bass guitar and piano, not singers or "seven drum beats."

> Singers aren't always mixed center, and snare tracks have no midbass. The CD pans pink noise (tracks 6–10), bass guitar (11–15, checks midbass delay and level) and piano (16–20, should spread wider) across five positions; half-left should land exactly halfway between center and far left. Track 25 is for the last-resort front delay, track 38 (silence) for noise. RARE **Conflict:** the attributed gain guide still suggests "7 drum beats" for centering.
>
> **Sources:** UMI-1 video @28:00 · Tuning PDF v1 (~2018) · DIYMA 2013

In plain English: Many people check their stage with a singer, or with a track of seven drum beats panned from left to right. He says singers aren't always mixed in the center, and snare drum tracks contain no midbass. Instead, his test CD pans three sounds across five positions from left to right: pink noise (tracks 6–10), bass guitar (tracks 11–15, which check the midbass delay and level) and piano (tracks 16–20, which should spread wider). The half-left track should land exactly halfway between the center and far left. Track 25 is for the last-resort front delay (see 3.9 #12), and track 38, which is silence, is for checking noise (see 3.10 #6). The tag notes a conflict: the gain guide attributed to him still suggests "7 drum beats" for centering.

3.11 #5 Listen with band-limited noise or music, never pure sine tones.

> Above 1 kHz the reflection comb is so dense that a tone easily lands in a null, and the nulls differ left to right; that is why 1980s tone-based centering (and "31-sine" tuning) fails, while music and head movement erase the comb. If you must tune by ear, use band-limited pink noise. RARE
>
> **Sources:** DIYMA 2017-07-12 · DIYMA 2021-07-08

In plain English: A sine tone is a single pure pitch, like a tuning fork. In a car, reflections create a pattern of dips (nulls) where sound cancels, packed very tightly above 1 kHz. A single tone can easily land in one of those nulls, and the nulls fall in different places for the left and right speakers, so a tone misleads you. That is why the 1980s method of centering with tones, and "31-sine"

tuning, fail. Music covers many frequencies at once, and we naturally move our heads, which erases the comb of dips. If you must tune by ear, use band-limited pink noise (hiss filtered to one range).

3.11 #6 Don't tune for depth or height; the stage comes from the recording and the car.

> Images fall inside the triangle of you and the speakers; depth and width come from the recording's ambience ("there's no trick" beyond it), and cabin reflections only spread images. A static ~3.5 kHz "height" bump works only while you're adjusting it, then it's just a response error. RARE
>
> **Sources:** Tuning PDF v2 (~2023) · DIYMA 2021-11-11 · UMI-1 video @31:05

In plain English: Many tuners try to make the stage seem deeper (farther away) or higher (up on the windshield) with EQ or delay tricks. He says not to tune for that. Images appear inside the triangle formed by you and the two speakers. Depth and width come from the ambience captured in the recording (the sense of the space it was recorded in), and "there's no trick" beyond that; reflections in the cabin only spread images out. A popular height trick is a fixed boost around 3.5 kHz. He says it works only while you're adjusting it; after that it's just an error in the frequency response.

3.11 #7 Missing midbass "snap" usually lives around 1–3.5 kHz, not in the midbass.

> Look near 1 kHz first; flat dynamics are often the mastering (a 6 dB crest factor). If door-midbass impact is truly missing, raise the midbass/sub crossover and work the 60–160 Hz EQ. RARE
>
> **Sources:** DIYMA 2015 · DIYMA 2011

In plain English: "Snap" or "punch" is the sharp, impactful attack of a kick drum or bass note. When it's missing, many people turn up the midbass. He says the missing snap usually lives around 1–3.5 kHz, not in the midbass, so look near 1 kHz first. Flat, lifeless dynamics are often down to the recording's mastering. "Crest factor" is the gap between a recording's loudest peaks and its average level; a crest factor of only 6 dB means heavily squashed music. If the impact from the door midbass speakers really is missing, raise the midbass/sub crossover frequency and work on the EQ between 60 and 160 Hz.

3.11 #8 How experienced listeners judge a car in seconds.

> Full, impactful bass that seems to come from the front; a centered singer; then width and depth, which partly belong to the recording.
>
> **Sources:** Absolute Electronix livestream @53:21

In plain English: He describes what experienced listeners check first when they sit in a car: bass that is full and impactful and seems to come from the front (not the trunk); a singer who sits in the center; and then the width and depth of the stage, which partly depend on the recording rather than the car.

3.11 #9 Listen to factory premium systems for what they do well.

> Aftermarket listeners judge them on bass and loudness and miss the envelopment and two-seat imaging OEM engineers built in. RARE
>
> **Sources:** Absolute Electronix livestream @56:27

In plain English: Factory premium systems are the branded sound systems carmakers offer as upgrades. OEM (original equipment manufacturer) means the carmaker and its suppliers; aftermarket means equipment installed later. He says aftermarket enthusiasts judge factory systems on bass and loudness, and so miss what OEM engineers built in: envelopment (the feeling of being surrounded by sound) and imaging that works for both front seats.

3.11 #10 Find your stage's true edges with the balance control.

> With delays and L/R EQ done, pan fully left: that is as far left as the system will image consistently; then fully right. Nothing in a simple stereo system, in a space about half the size of a bathroom, reliably puts images far outside the speakers. RARE
>
> **Sources:** DIYMA 2017-08-18

In plain English: Many people chase a stage that stretches beyond the car's doors. He gives a simple test of the real limits. Once the delays and the left/right EQ match are done, turn the balance control fully left: wherever the sound appears is as far left as your system will place images consistently. Then do the same fully right. He says nothing in a simple stereo system, in a space about half the size of a bathroom, reliably puts images far outside the speakers.

3.12 MS-8 auto-EQ, processors and DSP features

What this part is about: The JBL MS-8 was a car-audio processor from his years at Harman that measured the car and tuned itself automatically (an "auto-EQ"). Most of this part relates to that specific product: how it measured and corrected, its quirks and setup facts. The last points cover other auto-tuning products and smaller DSP features.

3.12 #1 MS-8 measures with a binaural headset at several head positions, at conversation level.

> Mics sit at both ears; up to four seats; four measurements per seat (forward twice, toward each mirror), holding still during sweeps. Per position: 3 sweeps × 2 mics = 6 response measurements; for delay, one sweep per channel × 2 mics. Wearable mics let you work the remote while measuring; calibrating too loud degrades results. Product documentation plus his posts.
>
> **Sources:** MS-8 manual (mirror) · DIYMA 2017-06-28

In plain English: This point relates to one product, the JBL MS-8. "Binaural" means two ears: the MS-8's measuring microphones sit on a headset, one at each ear, so it measures roughly what your ears hear. It can calibrate for up to four seats. For each seat it takes four measurements with your head in different positions (facing forward twice, then turned toward each side mirror), and you hold still while it plays its sweeps (tones gliding from low to high pitch). At each position, 3 sweeps × 2 mics gives 6 response measurements; for setting delay it plays one sweep per channel, picked up by both mics. Because the mics are worn, you can work the remote while measuring. He says calibrating too loud makes the results worse; it measures at conversation level. This point draws on the product's documentation plus his own posts.

3.12 #2 How MS-8's auto-EQ really worked: a correction built from a spatial average.

> It took six impulse measurements, converted each to a response (FFT), averaged them, built a minimum-phase version (an average has no phase), converted back to an impulse, and applied 8 filters per channel to successively longer slices of it, so a filter could take almost any shape; the result was convolved with the audio. Corrections apply per speaker location, not per crossover channel. Ulrich Horbach designed it, not Andy. RARE **Conflict:** his 2011 FAQ described 8 biquad filters per channel that "can do more work than 512 taps"; his 2017 and 2022 descriptions are FIR-like (filters applied to the impulse, then convolution).
>
> **Sources:** DIYMA 2017-06-28 · DIYMA 2022-02-03 · DIYMA MS-8 FAQ 2011

In plain English: A "spatial average" combines measurements from several positions around the listener's head, so the correction suits the space your head occupies rather than one exact point. Step by step, as described here: the MS-8 took six impulse measurements (see 3.9 #5); turned each into a frequency response using an FFT (a standard calculation that converts a recording over time into levels at each frequency); averaged them; built a "minimum-phase" version, which assigns the most compact possible timing to that averaged response, needed because an average has no phase of its own; turned that back into an impulse; and applied 8 filters per channel to successively longer slices of it, so each correction could take almost any shape. The result was convolved with the audio, meaning applied to the music as one combined filter. Corrections apply per speaker location (for example, the left front), not per crossover channel (for example, just its tweeter). He notes that Ulrich Horbach designed it, not he. The tag flags a conflict: his 2011 FAQ described 8 "biquad" filters per channel (the basic building block of most digital EQs) that "can do more work than 512 taps" (a measure of FIR filter size), while his 2017 and 2022 descriptions sound more like FIR filtering: filters applied to the impulse, then convolution.

3.12 #3 With an auto-EQ, crossover settings are only suggestions; limit each channel's EQ to its passband.

> MS-8's EQ combines with your filters to set the final acoustic slopes, so choose crossovers for safety and output and recalibrate after any change. In other auto-EQs, using the full-range target on every channel makes the EQ fight your crossover slopes. RARE
>
> **Sources:** DIYMA MS-8 FAQ 2011 · DIYMA 2021-04-17

In plain English: Normally you'd think the crossover setting alone decides where a speaker stops playing. With the MS-8, he says, the auto-EQ adds its own corrections on top of your crossover filters, and the two together set the final slopes you actually hear. So the crossover settings are only suggestions: choose them for the speaker's safety and for the output you need, and recalibrate after any change. With other auto-EQ systems, limit each channel's EQ to its passband (the range it is meant to play). If you give every channel the full-range target curve, the EQ fights your crossover slopes, trying to boost back what the crossover cut.

3.12 #4 MS-8 sets sub level from 50–80 Hz before EQ, and that can over-boost the lows.

> A 15" in 0.5 cu ft crossed at 40 Hz plays much louder above 40 Hz and fools it; a small sealed box peaking at 50–80 Hz gets turned down. Fixes he gives: a 50 Hz 12 dB low-pass on the sub amp, the 31-band EQ, or (2018) reverse the sub's polarity and set the sub amp's crossover to its highest setting. He'd have changed this algorithm. RARE
>
> **Sources:** DIYMA MS-8 FAQ · DIYMA 2018-04-08

In plain English: This is an MS-8 quirk. Before it EQs, the MS-8 sets the subwoofer's overall level from how loud the sub measures between 50 and 80 Hz. That can fool it. His examples: a 15-inch sub in a tiny 0.5 cubic foot (about 14 liter) box, crossed at 40 Hz, still plays much louder above 40 Hz than below, so the MS-8 sets the bass level from that louder 50–80 Hz output and then tries to EQ the deep bass back in; and a small sealed box whose output peaks at 50–80 Hz gets turned down. Either way, the result can be over-boosted lows. The fixes he gives: a 50 Hz low-pass filter with a 12 dB per octave slope on the sub amplifier; the MS-8's 31-band EQ; or, in 2018, reversing the sub's polarity and setting the sub amp's crossover to its highest setting. He says he would have changed this algorithm.

3.12 #5 MS-8 delay and seat modes.

> It averages one impulse from both ear mics, takes the first highest peak, backs up 12 dB and delays every channel to the latest; fronts are not aligned to subs. "Passenger" reverses the delays; "Front/center" removes L/R delays and keeps only crossover-matching ones; "Driver" beats "Front" only in placing midbass below the center's high-pass.
>
> **Sources:** DIYMA 2013 · DIYMA 2018-04-07 · DIYMA 2018-11-16

In plain English: For timing, the MS-8 averages one impulse measurement from the two ear mics, takes the first highest peak, backs up to the point 12 dB below it to estimate the arrival, and delays every channel to match the latest-arriving one. It does not align the front speakers to the subs (compare 3.9 #12). Its seat modes: "Passenger" reverses the delays, mirroring the driver's setting. "Front/center" removes the left/right delays and keeps only the delays that line up speakers sharing a crossover. "Driver" beats "Front" in only one way: placing the midbass below the center speaker's high-pass (the frequency below which the center stops playing). He adds that "Front" leaves the midbass leaning toward the nearer side.

3.12 #6 Check an auto-EQ with the same kind of measurement it used.

> MS-8's head-worn mics give a darker target than a single mic, so a single-mic check won't match; use a multi-position average. Don't add outside correction to its sides and rears (they can't be encoded back into stereo); if you need per-driver delay, put a MiniDSP after MS-8, set its delays, then calibrate. RARE
>
> **Sources:** DIYMA 2017-08-05 · DIYMA 2018-04-08 · DIYMA 2011

In plain English: If you check an auto-EQ's result with your own microphone, measure the way it did. The MS-8's mics are worn on your head, so its target comes out "darker" (less treble) than a single-mic measurement would show. A single-mic check therefore won't match what it did; an average of measurements from several positions can confirm it. Don't add your own outside correction to the MS-8's side and rear channels; he says those can't be encoded back into stereo for such correction. If you need separate delays for each driver, put a MiniDSP (a small, inexpensive DSP) after the MS-8, set its delays first, then run the MS-8 calibration.

3.12 #7 MS-8 setup and service facts.

> With a factory radio, learn whether it is fed from the radio or the factory amp; with an aftermarket radio use low-level inputs (fronts to inputs 1–2) and skip input calibration. It may not boot in cold weather because it waits for its slow LCD (its display has its own processor, so it can't run without

> it). No real custom firmware exists. Random balance shifts usually mean an intermittent input.
>
> **Sources:** DIYMA 2021-12-28 · DIYMA 2022-01-05 · DIYMA 2019-01-13

In plain English: A few practical MS-8 facts. With a factory radio, find out whether the MS-8 is being fed from the radio itself or from the factory amplifier. With an aftermarket radio, use the low-level (line-level) inputs, with the front channels going to inputs 1 and 2, and skip the input calibration step. In cold weather it may not start up, because it waits for its slow LCD screen; the display has its own processor, so the unit can't run without it. No real custom firmware (unofficial replacement software) exists for it. If the balance shifts randomly, that usually means an input with an intermittent (on-and-off) connection.

3.12 #8 MS-8's purpose and what he'd change.

> He hoped it would end the "mythology" that tuning is an art, and stopped wanting a manual back door. In 2016 he'd have replaced L/R delay with phase EQ for two seats, changed the sub-level algorithm, and deepened the 2 kHz dip in the center-channel target. To put the image "on the hood": cut some 500 Hz–2 kHz and adjust center level.
>
> **Sources:** DIYMA 2010 · DIYMA 2016 · DIYMA MS-8 FAQ

In plain English: He hoped the MS-8 would end the "mythology" that tuning is an art, by doing it through measurement. Early on he asked its engineers for a manual "back door" for hand adjustment, but later stopped wanting one. Looking back in 2016, he would have changed three things: replaced left/right delay with phase EQ (all-pass filters, see 3.8 #14–15) so both front seats work; changed the sub-level algorithm (see #4 above); and made the 2 kHz dip in the center channel's target curve deeper. "On the hood" means an image that seems to sit out beyond the windshield. To get it there: cut some energy between 500 Hz and 2 kHz and adjust the center channel's level. He treats an image "on the hood" as a good result to aim for.

3.12 #9 Auto-EQs need phase, so they build a minimum-phase curve and must cap Q and boost.

> That hides regions that aren't minimum phase, so the algorithm protects itself against boosting into nulls. Hand workaround for tools that need phase: paste the phase of a single-mic measurement that resembles the average into the average's file. Averaging phase itself is meaningless (−180° and 0° "average" to −90° but still null). RARE
>
> **Sources:** DIYMA 2017-06-27 · DIYMA 2017-06-28 · DIYMA 2017-06-27

In plain English: To design accurate filters, most auto-EQs need phase (timing) information as well as levels. A spatial average has no phase, so they build a minimum-phase version of it (see #2). That hides the parts of the response that aren't minimum phase. Background: deep, narrow dips caused by cancellation (nulls) can't be filled by boosting, because the added sound cancels too. So the software protects itself by capping Q (how narrow a filter can be) and how much it can boost, so it doesn't boost into nulls. His hand workaround for tools that need phase: take a single-mic measurement whose curve looks like the average, and paste its phase data into the average's file. Averaging phase itself is meaningless: two speakers at −180° and 0° "average" to −90°, yet they are still 180° apart and cancel.

3.12 #10 MS-2, Dirac and APL1 in brief.

> After MS-8, Harman found a smoothed single-mic average about 90% as accurate as a spatial average, so MS-2 used one mic and two 512-tap FIRs split at 1 kHz (~1.9 Hz resolution below it). Dirac uses IIR for response plus an FIR working on phase, which improves imaging more than delays can (he hasn't used it and won't rank it). APL1 averages the whole cabin and applies a 4096-tap FIR; repeatability is one test of any automated process.
>
> **Sources:** DIYMA 2017-06-26 · DIYMA 2022-02-03 · DIYMA 2017

In plain English: This point covers three other auto-tuning products. Background: FIR (finite impulse response) and IIR (infinite impulse response) are two kinds of digital filter; FIR filters can shape timing (phase) as well as level, and more "taps" means finer detail. MS-2 was a later Harman processor. After the MS-8, Harman found that a single-mic measurement, smoothed (averaged across neighboring frequencies), was about 90% as accurate as a spatial average, so the MS-2 used one mic. It used two 512-tap FIR filters split at 1 kHz, giving about 1.9 Hz resolution below 1 kHz. Dirac (a commercial correction system) uses IIR filters for the response plus an FIR working on phase, which he says improves imaging more than delays can; he hasn't used it and won't rank it. APL1 averages measurements across the whole cabin and applies a 4096-tap FIR. One test of any automated process, he notes, is repeatability: does it give the same result each time?

3.12 #11 Check an upmixer with pink noise.

> In Logic 7/Pro Logic II, mono noise should come mostly from the center (fronts about 6 dB down); mono noise with one channel inverted comes from the rears; balance hard right plays only the right front; stereo noise exercises everything and is the best overall check. Most DSP makers omit upmixers not for license cost but because product managers don't understand them or judged them from bad setups. RARE
>
> **Sources:** DIYMA 2017-08-05 · DIYMA 2017-07-07

In plain English: An upmixer, such as Logic 7 or Dolby Pro Logic II, turns two-channel stereo into more channels: center, sides and rears. His pink-noise checks: mono noise (identical in both channels) should come mostly from the center speaker, with the left and right fronts about 6 dB quieter. Mono noise with one channel's polarity inverted (so the two channels are exact opposites) comes from the rear speakers. With the balance hard right, only the right front plays. Stereo noise (different in each channel) exercises every speaker and is, in his view, the best overall check. He adds that most DSP makers leave upmixers out not because of license costs, but because their product managers don't understand them or judged them from badly set-up systems.

3.12 #12 Tuning with a binaural head is harder, not easier.

> A dummy head's response changes with every direction, so you need either an average that is exact for no direction or a separate calibration file per speaker location (dash, door, center). RARE
>
> **Sources:** Dean & Fernando live @58:03

In plain English: A binaural head (dummy head) is a model head with microphones in its ears, meant to measure what a person hears. You might expect it to make tuning easier. He says it makes it harder. A head and ears change the measured response depending on which direction the sound comes from. So you need either an average correction, which is exactly right for no direction, or a separate calibration file for each speaker location (dash, door, center).

3.12 #13 Smaller DSP notes.

> Test claimed "transient" differences between DSPs with a null test (align recordings, invert one, mix; what's left is the difference). Filter arithmetic precision (single vs double, fixed vs floating point) matters. Head-unit control of a Helix fails if the head unit's Wi-Fi is joined to an in-car hotspot.
>
> **Sources:** DIYMA 2021-01-04 · DIYMA 2021-12-18 · DIYMA 2021-09-02

In plain English: Three short notes. First, some people claim one DSP handles "transients" (sudden, sharp sounds such as drum hits) better than another. He says test such claims with a null test: record the same music through both, line the recordings up exactly, invert one, and mix them; whatever is left over is the real difference. Second, the precision of the arithmetic a DSP uses for its filters matters: "single vs double" precision and "fixed vs floating point" are different ways of handling numbers, with different accuracy. Third, Helix is a DSP brand that some head units (car radios) can control; that control fails if the head unit's Wi-Fi is joined to an in-car hotspot.

3.13 Tuning mistakes he sees most

What this part is about: This part lists the tuning errors he says he runs into most often. Many of them point back to rules explained in earlier parts (phase, delay, gain setting, crossovers), and each one shows what goes wrong when a rule is skipped or mixed up.

3.13 #1 Fixing the center by flipping one mid, then adding delays on top.

> Result: midbass 180° out at all frequencies and no center (3.8 #2).
>
> **Sources:** Tuning PDF v2 (~2023) · OFCAP Ep. 19 @3:17:02

In plain English: The "center" is the illusion of a singer standing in the middle of the dashboard, even though the sound comes from left and right speakers. People use two separate fixes for a weak center. One is flipping the polarity of one front speaker (making its cone move backward when the other moves forward). The other is delay: holding back the sound from the nearer speakers so everything arrives at your ears together. He says the mistake is doing both. His point is that you can do one or the other, not both. Once the delays line up the arrival times, the flipped speaker is exactly opposite to the other one, 180° out at every frequency. In the midbass the two sides then cancel, and the center is gone.

3.13 #2 Entering delays wrongly, then blaming the method.

> A competition car with great bass had no center because the owner typed distances into a DSP that wanted delays; "the tape measure doesn't work" is usually this. His rule: if you follow the manual and don't get the result, assume you made an error and keep looking. Forum "14-year breakthroughs" after endless rebuilds usually mean the delays were finally right. RARE
>
> **Sources:** Absolute Electronix livestream @44:23 · Absolute Electronix livestream @47:54 · OFCAP Ep. 19 @3:17:02

In plain English: Time alignment means delaying the nearer speakers so that every speaker's sound reaches your ears at the same moment. The usual way is to measure each speaker's distance to your head with a tape measure. Different DSPs (digital sound processors, the box that does the tuning) want that number entered in different ways: as a time, as a distance, or as each speaker's own distance, which you then have to subtract. He describes a competition car with great bass but

no center, because the owner typed distances into a DSP that wanted delays. He says that when people claim "the tape measure doesn't work", it is usually this kind of entry mistake. His rule: if you follow the manual and don't get the result, assume you made an error and keep looking. Many people believe their big breakthrough came from years of rebuilding panels and moving speakers. He says those forum "14-year breakthroughs" usually mean the delays were finally set right.

3.13 #3 Using delay as a pan pot, a depth knob or a crossover-phase fix, or aligning the fronts to the sub.

> Delay corrects distance and nothing else (3.9 #1, #8, #9, #12).
>
> **Sources:** Audiofrog blog Dec 2020, Time Alignment Part 1 · DIYMA 2015 · DIYMA 2013

In plain English: A pan pot is a control that slides a sound left or right. A "depth knob" here means trying to push the stage farther forward or back. A crossover splits the music by frequency between speakers, and its filters shift the timing (phase) of the sound near the split point. He says delay has one job: making up for the different distances from each speaker to your ears. Using it to shift the image sideways, to add depth, to cure phase at a crossover, or to line the front speakers up with the subwoofer are all mistakes on his list. The points it refers to explain why (3.9 #1, #8, #9, #12). Note that his advice on delaying the sub itself changed over the years (see 3.9 #12 and 4.2 #12).

3.13 #4 Listening and tweaking by ear before the objective setup is done, or mixing several teachers' steps.

> By-ear tuning means aimlessly fiddling "for 20 minutes"; an amalgam of processes isn't a process ("ignorance is not a process"). Pick one that ends at correct, is teachable and fast, and practise it.
>
> **Sources:** DIYMA 2021-05-12 · OFCAP Ep. 19 @3:19:02 · Audiofrog blog 2021-04-14, UMi-1 CD

In plain English: Tuning has a measured part, done with a microphone (levels, delays, EQ toward a target), and a listening part. He says listening and tweaking by ear before the measured setup is finished turns into aimless fiddling "for 20 minutes". He also says that taking a few steps from one teacher and a few from another doesn't give you a process ("ignorance is not a process"). His advice: pick one process that ends at a correct result, can be taught, and is fast, and then practise it. Elsewhere he says to listen at the end to check the result, not to tweak by ear at every step.

3.13 #5 Chasing what the mic sees but you don't hear.

> EQing the summed response, boosting nulls, and fixing narrow, position-dependent dips (above ~1 kHz, Q above ~6). Move the mic a little: if a ripple stays, fixing it may help; if not, fixing it makes things worse. RARE
>
> **Sources:** Tuning PDF v1 (~2018) · Audiofrog blog 2021-01-14, single mic or multi-mic · DIYMA 2021-03-26

In plain English: A measurement mic shows many bumps and dips that your ears don't hear the same way. An EQ (equalizer) raises or lowers chosen frequencies. A null is a deep dip where two sound waves cancel. Q describes how narrow an EQ adjustment is: the higher the Q, the narrower. He lists these as mistakes: EQing the summed response (both sides measured playing together), boosting nulls, and fixing narrow dips that change when the mic moves (above about 1 kHz, with a

Q above about 6). Many people believe every wiggle on the graph should be flattened. His test is simple: move the mic a little. If a ripple stays put, fixing it may help. If it moves or disappears, it isn't a real feature of the sound at your head, and fixing it makes things worse.

3.13 #6 Clipping the DSP's input, then blaming the speakers.

> Setting the input range below the source voltage (a 1 V setting on a 4–5 V source allows ~12 dB of A/D clipping) causes permanent distortion that "sounds like broken speakers." He estimates ~70% of GB25 (2.5" midrange) warranty claims are DSP clipping from gain-setting errors; raising the tweeter crossover to "fix distortion" usually hides it (or a leaky pod). Setting gains with sines and a scope before tuning is the related error. RARE
>
> **Sources:** Audiofrog blog 2021-03-24, scopes and sine waves · DIYMA 2022-03-13 · Audiofrog blog Dec 2020, pink noise

In plain English: A DSP turns the incoming signal into numbers with an A/D (analog-to-digital) converter. Its input range setting (for example 1 V or 4 V) must match how strong the source's signal is. Set it to 1 V when the radio puts out 4–5 V, and the tops of the waves get chopped off; this is "clipping", by about 12 dB in his example. Because it happens before the processing, the distortion is permanent and nothing later can remove it. It "sounds like broken speakers." He estimates that about 70% of warranty claims on his GB25 (a 2.5-inch midrange) are really DSP clipping from gain-setting errors. People often raise the tweeter's crossover point to "fix" the distortion. He says that usually just hides it, or hides a leaky pod (the small speaker housing not sealed properly). He also calls setting gains with sine-wave tones and an oscilloscope before tuning a related error. So in practice: set the input range to suit your source first, so the DSP's input can't clip, even at full volume.

3.13 #7 Crossovers by wishful thinking.

> Asking a door midbass for 40–60 Hz, wide sub/midbass overlap, mids below Fs, tweeters below ~1.5× Fs, or a steep slope used to justify a low tweeter point. People also mistake a big peak for the speaker's real output: set the target near the dips and cut the peaks instead.
>
> **Sources:** DIYMA 2010 · Audiofrog blog 2021-10-07, slopes and protection · DIYMA 2021-07-12

In plain English: Fs is a speaker's resonant frequency, the pitch where it naturally vibrates most easily; below it, its output falls away. Here he lists crossover choices made on hope rather than on what the speakers can do: asking a door midbass to play deep bass at 40–60 Hz; letting the sub and midbass overlap over a wide range; playing midranges below their Fs; crossing tweeters lower than about 1.5 times their Fs; and counting on a steep slope (a filter that cuts off sharply) to justify crossing a tweeter low. He also says people mistake a big peak on the graph for the speaker's real output level. Instead, set the target level near the dips and cut the peaks down to it.

3.13 #8 Aiming midranges to fix reflection dips, or boosting a dash speaker's natural treble roll-off.

> **Sources:** Audiofrog blog 2021-06-14, dispersion and reflection

In plain English: Two more mistakes. First, aiming midranges to try to fix the dips caused by reflections (sound bouncing off glass or panels and arriving a moment after the direct sound). In the source article he explains that a midrange sends sound in all directions below a certain frequency, so aiming it does nothing to its reflections; only a tweeter's very top end can be steered

by aiming. Second, boosting the treble that a dash speaker naturally loses. He says a dash speaker's steep treble roll-off (its fall in output toward the high notes) shouldn't be treated as a dip to fill: treat what lies below it as a peak and cut that instead.

3.13 #9 Arguing over whose target curve is best instead of hitting one; or "correcting" a car with a 31-band RTA and 31-band EQ.

> **Sources:** Audiofrog blog 2021-03-10, target curves · DIYMA 2008

In plain English: A target curve is the shape you tune the frequency response toward, for example more bass and gently falling treble. He says people waste time arguing about whose target is best, when the real work is actually hitting one. An RTA (real-time analyzer) shows sound level in bands; a 31-band graphic EQ has 31 fixed sliders. He says this combination's resolution is too coarse to correct a car, though a 31-band EQ is passable for shaping a curve that has already been corrected properly.

3.13 #10 Ignoring the factory processing already in the signal, or insisting on "flat" first.

> Factory all-pass filters can't be "un-EQ'd," so getting "back to zero" first is often inefficient (3.8 #20), and leftover factory delay wrecks yours (3.9 #13). In upmixing factory systems he often keeps the processing: map each factory output to a matching amp channel, add power and better speakers, re-EQ and set levels, with no new delays or crossovers; ripping out factory locations is "a big mistake." RARE
>
> **Sources:** Absolute Electronix livestream @58:34 · CMA Connected 2021 @37:25 · CMA Connected 2021 @38:54

In plain English: Factory systems often process the signal before it reaches the speakers: EQ, delays and all-pass filters. An all-pass filter shifts the timing (phase) of a band of frequencies without changing its level, so an EQ cannot undo it. That is why he says trying to get "back to zero" (a flat, unprocessed signal) before tuning is often an inefficient way to work. He also says leftover factory delay will spoil any delays you add on top. Many people believe the first step is always to flatten the factory signal. In factory systems that "upmix" (spread stereo music over many speakers), he often keeps the factory processing. He connects each factory output to a matching amp channel, adds power and better speakers, re-EQs and sets levels, with no new delays or crossovers. He calls ripping out the factory speaker locations "a big mistake."

3.13 #11 Four months of fabrication, 20 minutes of tuning.

> Shops show him beautiful builds that don't sound great and say "not bad for the 20 minutes I spent"; the audio deserves real time. RARE
>
> **Sources:** CMA Connected 2021 @22:22

In plain English: Fabrication means custom-building panels, pods and boxes. He says shops show him beautiful builds that don't sound great, with remarks like "not bad for the 20 minutes I spent" on tuning. His point: if the build took months, the sound deserves real tuning time too.

3.13 #12 Treating customers who don't care about imaging as not needing a DSP.

> Club-style listeners need it for driver safety and for midbass-to-sub phase, which is where impact

> comes from (3.8 #10).
>
> **Sources:** CMA Connected 2021 @46:57

In plain English: Imaging is hearing each instrument at its own spot on a stage in front of you. Some customers only want loud, punchy, club-style sound and don't care about that. He says they still need a DSP, for two reasons: keeping the speakers (drivers) safe, and getting the midbass and subwoofer in phase, which he says is where impact comes from (3.8 #10). Note that 4.2 #3 records a different statement of his about where "impact" comes from.

3.13 #13 Tearing an install apart for hiss that the auto-EQ created.

> If a factory head unit rolls off the highs, an auto-EQ boosts the top end to compensate, and that boost is the hiss; it isn't a ground fault. RARE
>
> **Sources:** DIYMA 2021-08-08

In plain English: Hiss is a steady background "shhh". Auto-EQ is a feature that measures the car and sets the EQ automatically. If the factory head unit (the radio) turns down the high frequencies, the auto-EQ boosts the top end to make up for it, and that boost is what you hear as hiss. He says it isn't a ground fault (a bad electrical connection to the car's body), so there is no point tearing the install apart looking for one. Many people assume hiss always means a grounding problem.

Top rarely-discussed points in 3.1–3.7

1. Measure from outside the car with the door shut; his target assumes an empty car (3.4 #1)
2. Never wave the mic around your head from the seat; your head eats the treble and no target allows for it (3.4 #2)
3. Matching left to right beats hitting the target; copy an unfixable dip to the other side (3.6 #2)
4. Slide the target onto the dips like a router template and cut the peaks down to it (3.6 #3)
5. "Does it respond to EQ?" separates fixable minimum-phase problems from reflection or two-speaker interference (3.6 #10)
6. The "Harman curve" began as his installer curve, and popular targets differ by only ~2 dB (3.5 #2, #3)
7. One mic with 1/6-octave smoothing gets about 90% of a spatial average; five or six positions are plenty (3.4 #5, #7)
8. Lower the sub low-pass half an octave for every 12 dB of sub boost, keeping 60/80 Hz LR4 (3.7 #9)
9. Bass "in front" is sub–midbass crossover phase; door midbass crossed at 40 Hz is a red flag (3.7 #10, #11)
10. Steeper slopes don't protect tweeters more; 48 dB/oct passes more power near the knee than LR4 (3.7 #16)
11. Swapping speakers is just a non-adjustable EQ; EQ is the main lever (3.2 #9)
12. A 250 Hz "resonance" is usually a cabin reflection to EQ, not a panel to deaden (3.6 #14)

Changed views and conflicts in 3.1–3.7

1. **Order of steps:** 2011 EQ-before-delay → delay first from 2013; 2015 "two valid orders"; 2018 rule: one EQ per channel = delay first, one EQ per driver = EQ to the crossover target first (3.1 evolution #2–#6).
2. **Starting gains:** gain guide −12 dB on all DSP channels → 2022 sub-amp clip point first, then −6 dB (3.1 #2).
3. **Where the target sits:** PDFs put the midrange at 0 dB; his 2021 video, forum posts and the gain guide put the target on the dips (3.1 #7, 3.6 #3).
4. **Sweeps vs RTA:** 2012 sweeps "much better" → a live pink-noise RTA for production from 2018 (3.3 #1).
5. **Uncorrelated noise:** 2014 suggested it for 31-band work → 2020–22 useless except for upmixers, L−R rears or gaming an RTA (3.3 #4).
6. **REW settings:** PDFs FFT 16384, rectangular window, "dB not dBFS" → 2021 32k, window and units don't matter (3.3 #5).
7. **Single mic vs spatial average:** 2013 one mic worse than an MS-8-style average → 2017–2021 a smoothed single mic is ~90% as good and fine for production (3.4 #5).
8. **Smoothing numbers:** 2012–13, 2017, 2020 and 2022 give different splits; the rule (coarser above ~1 kHz) holds (3.4 #11).
9. **Target numbers:** +6–9 dB bass (2008) → +9 dB (MS-8, 2011) → +10 dB (2018 PDF); the shape never changed (3.5 #4).
10. **Narrow EQ:** blog says avoid Q above ~6; both PDFs still show a Q 10 cut at 1.6 kHz (3.6 #7).
11. **Tweeter floor:** 2009 "2× Fs, ~3 kHz for 1-in tweeters" vs PDFs 1.5–2× Fs at 2.2–2.5 kHz; 2022 forum puts the Audiofrog tweeter at 3 kHz (3.7 #2, #7).
12. **Sub/midbass points:** MS-8 era same point (80/80) → half-octave offset (60/80) from 2018 (3.7 #9).
13. **Seats:** one-seat PDF process → two-seat tuning where tools allow (2016 onward), but he won't teach all-pass two-seat tunes as a process (3.2 #13).
14. **Focus:** by 2023 he moved from teaching tuning to teaching system design by mapping the car (3.2 #15).
15. **Tweeter capacitor:** a stated precondition in PDF v1, not restated in v2, still recommended in his 2017 forum posts and Dec 2020 blog (3.1 #1, 3.7 #18).

Top rarely-discussed points in 3.8–3.13

1. An all-pass on one side fixes the ~250 Hz center problem for both front seats; delay fixes one and doubles the other's error (3.8 #14, 3.9 #14)
2. Two quick tests reveal factory all-pass vs delay: a single 200–500 Hz dip in L+R vs a comb (3.8 #17)
3. Keep factory polarity and add no delays on top of factory all-pass filters; they can't be inverted (3.8 #20)
4. Read phase from the L+R sum: +6 dB in phase, +3 dB ≈ 90°, equal to one side ≈ 120° (3.8 #7)

5. In-phase crossover pairs are heard at the higher-frequency driver; that's how bass comes forward (3.8 #10)
6. Delay doesn't move speakers; center with level, never delay (3.9 #1, 3.10 #2)
7. Impulse peaks aren't arrival times for subs and midbass; a tape measure is as good or better (3.9 #5)
8. Auto time alignment measures dash speakers via the windshield (3.9 #6)
9. Reflections and cabin gain never change delay settings (3.9 #10)
10. Learn imaging from a matched $5 thrift-store pair; "in phase" is the first reference (3.11 #2)
11. Seven-drum-beat and vocal tracks can't check midbass; use panned bass guitar (3.11 #4)
12. ~70% of GB25 (2.5" midrange) warranty claims are DSP input clipping from gain-setting errors, not bad drivers (3.13 #6)

Changed views and conflicts in 3.8–3.13

- **All-pass phase EQ** (3.8 #15): 2010, never got one to work for sub/midbass blending; 2014–15, Harman tests found "no audible difference"; from 2016, his preferred two-seat tool, and by 2021–22 the reason not to add delays to factory systems.
- **Aligning fronts to the sub** (3.9 #12): he argued for it at Harman, then accepted trials showing no difference below 100 Hz; the Tuning PDFs still tape-measure the sub like any speaker.
- **The passenger seat** (3.9 #14): once a trade-off against the driver's seat while delay was the only tool; now he calls "forget the passenger" a cop-out, since all-pass filters, placement or a center fix both seats.
- **Rear fill** (3.9 #18): 2017 recipes for differential L–R fill (delayed 12–20 ms); from late 2021, plain in-phase stereo delayed 10–20 ms.
- **MS-8 filter type** (3.12 #2): 2011 "8 biquads"; 2017 and 2022 an FIR-like correction convolved with the audio.
- **Test-track numbering** (3.11 #3–4): the 2021 UMI-1 video and the Tuning PDFs number the band-limited noise tracks differently.
- **"7 drum beats"** (3.11 #4): he says it can't check midbass; the attributed gain guide suggests it for centering.

SECTION 4

4. Bass, power and factory systems

Andy Wehmeyer treats bass as the easy part of car audio and bass that sounds like it comes from the front as the hard part: it depends on the car's cabin gain, a correct sub-to-midbass crossover and freedom from rattles and distortion. On power and gain he argues that music needs far less power than people buy, and that digital clipping, not too little power, is what kills speakers. On factory systems he says the OEM processing was built by experts, and that you should measure it and work with it before ripping it out.

What this section is about: This section covers three topics. The first is bass: the deep notes played by the subwoofer (a large speaker for the lowest notes) and the midbass speakers (usually in the doors). The second is amplifier power and gain settings. The third is factory-installed ("OEM") systems. In short, he says making bass in a car is easy, but bass that seems to come from in front of you is hard. On power, he says music needs far less power than people buy, and that digital clipping (waveforms chopped off inside the digital processing), not too little power, is what kills speakers. And he says factory systems were built by experts and should be measured and worked with before anyone rips them out.

4.1 How bass behaves in a car: cabin gain and the transfer function

What this part is about: This part explains how a car's cabin adds a lot of deep bass on its own (called "cabin gain"), how to measure that boost, and what it means when you choose a subwoofer box and tune it.

4.1 #1 The car boosts bass more and more as frequency falls; that boost belongs to the car, not the woofer.

> Cabin gain (the car's transfer function: in-car response divided by the woofer's outdoor response) keeps rising below 20 Hz, usually broken by a cancellation between 50 and 120 Hz. The in-car peak is where the box's roll-off outruns that gain, not a "car resonance"; an 8" and a 15" with the same outdoor response measure the same in the car. RARE
>
> **Sources:** DIYMA 2012-01-31 · DIYMA 2008-12-06

In plain English: Frequency, in hertz (Hz), is how many times per second a sound wave vibrates; the lower the number, the deeper the note. About 20 Hz is near the bottom of human hearing. He calls the car's effect on bass its "transfer function": the woofer's response measured in the car, divided by its response measured outdoors. That difference is the cabin gain. He says it keeps rising as the pitch falls, even below 20 Hz, and is usually interrupted by a cancellation dip somewhere between 50 and 120 Hz. Many people believe the bass peak they measure in a car is a "car resonance". He says it isn't: it is simply where the box's natural loss of deep bass (its roll-off) starts to outrun the car's boost. Think of the car as a fixed tone control that is the same whatever woofer you put in. That's why, he says, an 8-inch and a 15-inch woofer with the same outdoor response measure the same in the car.

4.1 #2 Every box type gets cabin gain; the car changes what you hear, not what the sub does.

> Sealed boxes look flat in cars only because their 12 dB/oct roll-off matches the rise; vented boxes stay flat lower and then fall faster, so they show a peak. RARE
>
> **Sources:** DIYMA 2009-08-28 · DIYMA 2015-06-07 · DIYMA 2021-05-08

In plain English: A sealed box is airtight. A vented (ported) box has an opening, the port, that adds output around one low "tuning" frequency. Below its limit, every box loses output at some rate; for a sealed box that is 12 dB per octave (an octave is a halving or doubling of frequency, and a few dB is a clearly audible change). He says every box type gets the same cabin gain. A sealed box only looks flat in a car because its 12 dB/oct loss happens to match the car's rise, so the two cancel. A vented box stays flat down to a lower pitch and then drops faster, so the car's rise shows up as a peak. Many people believe one box type suits cabin gain better. His point is that the car changes what you hear, not what the sub itself does.

4.1 #3 Where the sub sits barely matters below 40–50 Hz but shapes everything above 50 Hz.

> In his tests and Harman's, moving a box around a car changed nothing below 40–50 Hz; above that, cancellations appear, and the first cabin mode can be predicted from the car's size. Overall level depends on car size (an MG vs a Suburban, not a Jetta vs a Passat). RARE
>
> **Sources:** DIYMA 2008-07-05 · DIYMA 2008-12-06

In plain English: He says that in his own tests and in Harman's (the big audio company where he worked), moving a sub box around a car changed nothing below 40–50 Hz. At those low pitches a sound wave is many feet long, longer than the cabin. Above about 50 Hz, placement matters: cancellations appear, where waves meet out of step. He adds that the first cabin "mode" (the first strong resonance of the cabin's air, which he says usually shows up as the first big dip above 50 Hz) can be predicted from the car's size. Overall bass level also depends on size, but only when cars differ a lot: a tiny MG sports car versus a big Suburban SUV, not a Jetta versus a Passat (two similar sedans).

4.1 #4 Measure cabin gain with a ground-plane test.

> Measure at the seat; then put the box outdoors on the ground, 20–30 ft from walls, with the mic on the ground 2 m away at the same drive level (equal to an anechoic 1 m test). The difference is the cabin gain; this works only if woofer and port share a baffle.
>
> **Sources:** DIYMA 2009-08-28 · DIYMA 2008-10-12

In plain English: This is his way to measure how much bass the car itself adds. Step one: measure the sub inside the car, with the microphone at the seat. Step two: take the same box outdoors and set it on the ground, 20–30 ft (about 6–9 m) from any walls. Put the mic on the ground 2 m (about 6.5 ft) away and drive the sub at the same level. He says this equals a 1 m test in an anechoic room (a special echo-free test room). Step three: the difference between the two measurements is the cabin gain. He notes one condition: it works only if the woofer and the port are on the same baffle (the same face of the box).

4.1 #5 "Bass is the easiest thing to make in cars. Bass up front is not."

> The real work is 60 Hz–1 kHz. The cabin is a leaky resonator: soft panels leak, the seat filters between cabin and trunk, and windows act as passive radiators; find resonances by recording a mic while you whack the glass and sheet metal. RARE
>
> **Sources:** DIYMA 2008-11-20 · DIYMA 2011-12-24

In plain English: The quote means that getting a lot of deep bass in a car is easy; making it sound like it comes from in front of you is not. He says the real work is in the 60 Hz–1 kHz range (midbass and lower midrange). He describes the cabin as a leaky resonator: a space that rings at certain pitches but also lets sound escape. Soft panels let bass leak out. The back seat acts as a filter between the cabin and the trunk. Windows act as passive radiators, panels that get pushed by the air pressure and give off sound of their own, a bit like a drum skin. To find where the car rings, record with a mic while you whack the glass and the sheet metal.

4.1 #6 Don't design the box or pick the woofer to be the car's inverse; build the most efficient box you can fit, then EQ.

> Matching a sub's F3 (−3 dB point) to the car is "a ridiculous crapshoot". FIR inversion can't cancel the car either: nulls can't be filled, big boosts lack power and excursion, and the fix holds at one head position.
>
> **Sources:** DIYMA 2008-06-24 · DIYMA 2013-01-30 · DIYMA 2013-02-04

In plain English: F3 is the frequency where a speaker's output has dropped 3 dB below normal. Some people try to pick a woofer or design a box whose bass loss exactly mirrors the car's bass boost, so the two cancel. He calls matching F3 to the car "a ridiculous crapshoot". His advice instead: build the most efficient box that fits, then use EQ. He adds that FIR inversion can't cancel the car either. (FIR is a kind of digital filter that can be set to the exact opposite of a measured response.) He gives three reasons: nulls (deep cancellation dips) can't be filled; big boosts need more power and cone movement (excursion) than you have; and the correction is right at only one head position.

4.1 #7 No EQ available? Match the sealed box's −3 dB point to the car's +3 dB point.

> That gives flat lows, but he prefers vented plus a cut of the in-car peak (about 12 dB near 45 Hz in his example); short on space, use a high-Xmax woofer and lots of power.
>
> **Sources:** DIYMA 2008-05-25

In plain English: This is a fallback for a system with no EQ, from a 2008 post. If you choose a sealed box whose −3 dB point (where its output has dropped 3 dB) sits at the frequency where the car's gain has risen 3 dB, the two cancel and the lows come out flat. Even so, he prefers a vented box with EQ cutting the in-car peak, about 12 dB near 45 Hz in his example. He gives the reason: lower distortion and better dynamics, because power is left over for loud peaks. If you are short on space, he suggests a high-Xmax woofer (one whose cone can travel far while staying clean) with lots of power.

4.1 #8 Cut the cabin-gain peak, but by less than the graph says.

> For a boomy sub he cuts 6–10 dB rather than the 15 dB measured. A low-tuned box "lacks" 60–80 Hz only next to its 30–50 Hz peak: cut the peak and raise amp gain, which also lowers excursion and distortion.
>
> **Sources:** DIYMA 2015-10-09 · DIYMA 2011-12-26

In plain English: EQ lets you cut the peak that cabin gain creates. He says to cut it, but by less than the measurement suggests: for a boomy sub he cuts 6–10 dB rather than the 15 dB the graph shows. The source doesn't spell out exactly why he stops short. He also says that a box tuned low (one whose port works at a very low pitch) only seems to "lack" 60–80 Hz because it is being compared with its own 30–50 Hz peak. His fix: cut the peak and turn the amp gain up. That evens things out and also reduces cone movement (excursion) and distortion.

4.1 #9 Open windows, sunroof or trunk turn the cabin from a leaky sealed box into a low-tuned vented one; EQ toward that sound instead of cutting holes.

> Only very low bass rises. Tune with everything closed and save a second preset if you like the open sound; sealing the sub box to the cabin reduces the trunk-open boost but not the window-open one. RARE
>
> **Sources:** DIYMA 2008-08-25 · DIYMA 2013-08-25 · DIYMA 2018-03-23

In plain English: With everything closed, he describes the cabin as a leaky sealed box. Opening a window, the sunroof or the trunk turns it into a vented box tuned very low, with the opening acting like a port. Only the deepest bass rises. Some people cut holes in the car to get that sound. He says to EQ toward it instead. Tune with everything closed, and if you like the open-window sound, save it as a second preset. He adds that sealing the sub box to the cabin (so it doesn't play into an open trunk) reduces the extra bass when the trunk opens, but not when a window opens.

4.1 #10 Convertibles get little cabin gain, so build a bandpass box and fire its port into the cabin.

> A soft top is acoustically transparent; as a young installer he routinely did this. RARE
>
> **Sources:** DIYMA 2021-10-30

In plain English: A convertible's soft top is acoustically transparent: sound passes straight through it, so the cabin adds little bass. His answer is a bandpass box. In a bandpass box the woofer is fully enclosed and all the sound comes out through a port, so you can point that port into the cabin. He says that as a young installer he did this routinely.

4.1 #11 Home-style auto-EQ logic fails in cars because of cabin gain.

> Even a 6" door speaker can measure nearly flat below 20 Hz in a car, so the first Alpine Imprint (Audyssey-based), which put the sub crossover at the fronts' measured −3 dB point, always landed at its minimum.
>
> **Sources:** DIYMA 2016-01-25

In plain English: Auto-EQ systems measure the speakers and set things up automatically. Audyssey is a well-known room-correction system for home theaters. Home logic says: put the subwoofer crossover where the main speakers' bass fades, their −3 dB point. In a car, cabin gain props up the bass so much that even a 6-inch door speaker can measure nearly flat below 20 Hz.

So the first Alpine Imprint (based on Audyssey), which used that home rule, always put the sub crossover at its lowest possible setting. His point: this home-style auto-EQ logic fails in cars because of cabin gain.

4.1 #12 One mic position is enough for bass.

> Five positions in a 7" circle around the head measured almost the same below 200 Hz; they diverge from 200 Hz to 1 kHz and by 25 dB or more above 1 kHz. Keep bass EQ Q below about 6. RARE
>
> **Sources:** Audiofrog blog Jan 2021

In plain English: When measuring, some people average many mic positions around the head. He measured five positions in a 7-inch (about 18 cm) circle around the head. Below 200 Hz they read almost the same; from 200 Hz to 1 kHz they started to differ; above 1 kHz they differed by 25 dB or more. So one mic position is enough for bass. He also says to keep bass EQ adjustments with a Q below about 6. (Q is how narrow an EQ adjustment is; a higher Q is narrower.)

4.1 #13 Road noise: quieter tires were his best-ever midbass upgrade, and EQ cannot remove noise.

> Noise is usually negligible above 200 Hz, but on pebble-paved I-10 (El Paso–Van Horn) it mattered up to 500 Hz, so he kept a preset for that road. EQ only lifts music above the noise; record the noise and look at it in Audacity. RARE
>
> **Sources:** DIYMA 2016-01-25 · DIYMA 2008-05-30

In plain English: He says road noise is usually negligible above 200 Hz, so it mostly drowns out the low notes, and quieter tires were his best-ever midbass upgrade. EQ cannot remove noise; it can only raise the music above it. Removing noise itself takes active noise cancellation, a separate technology. On one rough, pebble-paved stretch of the I-10 highway (El Paso to Van Horn, Texas), the noise mattered up to 500 Hz, so he kept a separate EQ preset just for that road. To see your own noise, record it and look at it in Audacity (a free audio editor).

4.2 Getting bass to sound like it comes from the front: the sub–midbass crossover, phase and delay

What this part is about: This part is about the hardest bass job in a car: making the bass sound as if it comes from the stage in front of you, not from the subwoofer in the trunk. It covers the crossover that hands the bass from the sub to the midbass (door) speakers, how their phase (the timing of their waves) must line up, and what delay can and can't do.

4.2 #1 Front bass needs a correct sub–midbass crossover and zero rattles, not delay tricks or very low midbass.

> Any rattle or high-frequency distortion from the sub gives its location away, and bass 20 dB louder than everything else can't sound like it's in front. From 2017 to the 2023 tuning PDF he gives two conditions: sub and midbass in phase at the crossover, and no distortion. RARE
>
> **Sources:** DIYMA 2013-01-03 · DIYMA 2017-08-17 · V2 PDF 2023

In plain English: The sub usually sits in the trunk, behind you. In a good system the bass still seems to come from the front. A crossover is the filter pair that hands the low notes to the sub and the next notes up to the midbass speakers. He says front bass doesn't come from delay tricks or from making the door speakers play very low. It needs a correct sub-to-midbass crossover and zero rattles. Deep bass on its own is hard to locate, but he says any rattle or higher-pitched distortion from the sub gives away where it is. Bass that is 20 dB louder than everything else (a very large difference) can't sound like it's in front either. From 2017 through his 2023 tuning PDF he names two conditions: the sub and midbass in phase at the crossover (their waves pushing together), and no distortion.

4.2 #2 Distortion, not arrival time, is what makes bass localize to the trunk or a door.

> Rear-sounding bass often improves when you turn it down, and door woofers driven near resonance become easy to find. Test it: add 2nd and 3rd harmonics to a low-passed track, remove the fundamental, and A/B it in the car. RARE
>
> **Sources:** DIYMA 2013-03-18 · DIYMA 2010-02-13

In plain English: To "localize" a sound is to tell where it comes from. Many people believe bass sounds like it's in the trunk because it arrives late. He says the cause is distortion: extra, higher-pitched tones the speaker adds that weren't in the music. A harmonic is one of those tones, at a whole-number multiple of the note: the 2nd harmonic of 40 Hz is 80 Hz, the 3rd is 120 Hz. He says rear-sounding bass often improves when you simply turn it down. Door woofers pushed down near their resonance (the low end of their range) also become easy to pinpoint. His test, step by step: take a track and keep only its bass (low-pass it); create 2nd and 3rd harmonics of that bass; remove the original bass notes (the "fundamental") so only the added distortion is left; mix that back into the track; then switch between the two versions (A/B) in the car and listen.

4.2 #3 Kick-drum "impact" comes from the midrange (around the 3 kHz mid/tweeter crossover), not from 40 Hz.

> With a crossover at or below ~80 Hz the sub carries no attack, so by the precedence effect (the first arrival sets location) it can sit almost anywhere unless boosted hard. Low-pass a punchy track at 70–80 Hz on headphones and the impact disappears. RARE **Conflict:** in 2021 he also told dealers that a club-goer's "impact" comes from correct midbass–sub phase.
>
> **Sources:** DIYMA 2009-12-02 · DIYMA 2021-09-10 · CMA Connected Oct 2021 @46:57

In plain English: "Impact" is the punch or snap of a kick drum. Many people believe it comes from deep bass around 40 Hz. He says it comes from much higher, around 3 kHz, near the midrange-to-tweeter crossover. He says that without that region right you won't hear the snap of toms or a bass player's thumb slap. The "attack" is the sharp start of a note. With the sub crossed at or below about 80 Hz, the sub carries none of it. Your brain judges where a sound comes from by whatever arrives first (the "precedence effect"). Because the sub carries no attack, it can sit almost anywhere without giving itself away, unless it is boosted hard. His test: low-pass a punchy track at 70–80 Hz (keep only what's below) and listen on headphones; the impact disappears. The [Conflict] tag: in 2021 he also told dealers that a club-goer's "impact" comes from correct midbass-to-sub phase (3.13 #12). These are two different statements about where impact comes from, and the sources don't reconcile them.

4.2 #4 Big sub/midbass overlap destroys the image; a sub low-pass under full-range mains is "a low-pass filter, not a crossover".

> Without the matching high-pass, the sub goes out of phase with the mains about an octave above a 12 dB/oct cutoff (half an octave at 24 dB/oct). Harman's six-woofer optimizer (the basis of JBL BassQ) gave good response but "no image in any seat". RARE
>
> **Sources:** DIYMA 2009-11-14 · DIYMA 2013-04-27 · DIYMA 2022-02-10

In plain English: "Overlap" means the sub and the midbass both playing the same range of notes. A low-pass filter lets only the lows through; a high-pass filter lets only the highs through. A crossover is a matched pair of the two. Filters also shift phase, and a real crossover works because the two halves' shifts are designed to fit together. "Mains" are the main front speakers. He says a big overlap destroys the image (the sense of instruments placed on a stage). If the mains play full range and only the sub is filtered, that is "a low-pass filter, not a crossover". Without the matching high-pass, the sub goes out of phase with the mains about an octave above the cutoff with a 12 dB/oct filter, or half an octave above with 24 dB/oct. For example, with an 80 Hz cutoff at 12 dB/oct, that is around 160 Hz. He also describes a Harman optimizer that used six woofers (the basis of the JBL BassQ product): it gave good response, but "no image in any seat".

4.2 #5 Integration is "95% EQ and 5% phase" and needs real parametric bands per channel.

> With separate EQ on left midbass, right midbass and sub, he can blend at any crossover point or slope if nothing rattles; a 31-band graphic on a sub gives about 6 usable bands. Feed the sub from one left and one right DSP output so you can tune each whole side, sub included. RARE
>
> **Sources:** DIYMA 2013-04-27 · DIYMA 2015-05-04 · DIYMA 2021-08-22

In plain English: "Integration" means blending the sub and midbass into one seamless bass. A parametric EQ band lets you choose the exact frequency, the amount of boost or cut, and the width (Q). A graphic EQ has fixed sliders at set frequencies. He says blending is "95% EQ and 5% phase". With separate parametric EQ on the left midbass, the right midbass and the sub, he can blend at any crossover point or slope, as long as nothing rattles. A 31-band graphic EQ used on a sub gives only about 6 usable bands. (A 31-band EQ spreads its sliders across the whole audible range, so only a handful fall in the sub's range.) His wiring tip: feed the sub from one left and one right DSP output. That way you can tune each whole side, sub included.

4.2 #6 Default crossover: sub low-pass 60 Hz and midbass high-pass 80 Hz, both LR4 (24 dB/oct Linkwitz-Riley), because you will turn the sub up.

> Raising the sub 9–12 dB to sound natural (or riding a bass knob) moves the acoustic crossover up and out of phase. His rule: lower the sub low-pass half an octave per 12 dB of sub boost so the acoustic crossover stays at 80 Hz, with slopes EQ'd to target. RARE **Changed view:** 2010–2015 default was ~80 Hz LR4 on both sides; 2020–2022 the 60/80 Hz stagger.
>
> **Sources:** DIYMA 2010-02-14 · DIYMA 2020-12-30 · DIYMA 2021-08-21

In plain English: LR4 (a 4th-order Linkwitz-Riley filter) is a common crossover type that cuts 24 dB per octave and is designed so the two halves add up smoothly. The "acoustic crossover" is where the sub's and the midbass's actual sound levels cross in the car, which can differ from the number typed into the DSP. His default: sub low-pass at 60 Hz and midbass high-pass at 80 Hz, both LR4. The reason for the gap is that you will turn the sub up. Raising the sub 9–12 dB so the bass sounds

natural (or riding a bass knob) pushes the real crossover point higher, where the two speakers are out of phase. His rule: for every 12 dB of sub boost, lower the sub's low-pass by half an octave, so that the acoustic crossover stays at 80 Hz. Half an octave below 80 Hz is about 57 Hz, close to his 60 Hz. This assumes the slopes have been EQ'd to the target. The [Changed view] tag: from 2010 to 2015 his default was about 80 Hz LR4 on both sides; from 2020 to 2022 it was the staggered 60/80 Hz setting.

4.2 #7 Or add the bass lift with a shelf or a shared 80 Hz boost, not by delay.

> His tuning guide uses a 12 dB/oct low shelf near 60 Hz, set to taste (2009: never above 160 Hz, on the midbass channels too). The 2023 PDF: with the sub low-pass half an octave below the midbass high-pass, one boost near 80 Hz on both sub and fronts puts the bass in front with no delay changes. RARE
>
> **Sources:** DIYMA 2009-11-14 · DIYMA 2021-03-26 · V2 PDF 2023

In plain English: "Bass lift" is the extra bass most listeners want on top of a neutral tune. A low shelf is an EQ setting that raises everything below a chosen frequency, like a step. He says to add the lift with EQ, not with delay. His tuning guide uses a 12 dB/oct low shelf near 60 Hz, set to taste. In 2009 he added: never set it above 160 Hz, and apply it to the midbass channels too. His 2023 PDF gives another way. With the sub's low-pass half an octave below the midbass's high-pass, one boost near 80 Hz on both the sub and the front speakers puts the bass in front, with no changes to the delays.

4.2 #8 His 2008 integration recipe.

> One amp on both woofers plus a parametric EQ (ideally separate L/R); measure the sub alone, low-passed "ridiculously high"; cross where the mids roll off (~80 Hz); set the sub about 9 dB above the mids/highs with a smooth slope down to ~160 Hz; flip sub polarity if needed; don't boost the midbass. Era: 2008
>
> **Sources:** DIYMA 2008-11-20

In plain English: This is an older, 2008 step-by-step recipe. Step 1: run both woofers from one amplifier, and add a parametric EQ, ideally one with separate left and right controls. Step 2: measure the sub on its own, with its low-pass set "ridiculously high". Step 3: cross over where the mids and highs naturally roll off, about 80 Hz. Step 4: set the sub about 9 dB louder than the mids and highs, with the level sloping smoothly down toward about 160 Hz. Step 5: flip the sub's polarity if needed (polarity means which way the cone moves; flipping it reverses that). Step 6: don't boost the midbass with EQ. The Era: 2008 tag means this is how he described it then; later points in this part (such as #6 and #7) show his newer settings.

4.2 #9 Sub/midbass phase only has to stay far from 180°, and each slope type has its own correct phase.

> At 60 Hz a 1 ft distance error is ~19°, and even 60° apart sums louder than either alone. LR4 sums flat at 0°; LR2 at 180° (flip polarity); 2nd-order Butterworth at 0° gives a +3 dB hump; 3rd-order Butterworth sums flat at 90°/270° in either polarity, so it offers no polarity check. No speaker needs slopes steeper than 24 dB/oct. RARE
>
> **Sources:** DIYMA 2018-01-21 · DIYMA 2021-08-21 · DIYMA 2021-12-03

In plain English: Phase, in degrees, says how far one wave is shifted against another. At 0° their peaks line up and they add; at 180° one pushes while the other pulls, and they cancel. He says the sub and midbass only need to stay far from 180°. At 60 Hz a 1-foot (30 cm) distance error shifts them only about 19°, and even 60° apart they still add up louder than either alone. Crossover filters come in types (Linkwitz-Riley "LR", Butterworth) and orders (2nd order cuts 12 dB/oct, 3rd 18, 4th 24). Each type has its own correct phase. LR4 sums flat when the two speakers are at 0°. LR2 halves come out 180° apart, so you flip one speaker's polarity. A 2nd-order Butterworth at 0° gives a +3 dB hump at the crossover. A 3rd-order Butterworth sums flat at 90° or 270° in either polarity, so flipping polarity won't reveal a wiring mistake: it offers no polarity check. He adds that no speaker needs slopes steeper than 24 dB/oct.

4.2 #10 A dip at 60–90 Hz is usually sub-vs-front phase or a cancellation, and EQ can't fill a cancellation.

> Diagnose with sub and fronts crossed at 90 Hz and rears off; flip sub polarity if a big dip appears. A null near 70 Hz needs a sub move, or a crossover below it with the midbass matched.
>
> **Sources:** DIYMA 2008-11-18 · DIYMA 2016-04-03

In plain English: A dip is a range of notes that comes out too quiet. He says a dip between 60 and 90 Hz is usually the sub and the front speakers out of phase with each other, or a cancellation (sound waves meeting out of step). EQ can't fill a cancellation; boosting only makes both cancelling waves louder. His diagnosis: cross the sub and the fronts at 90 Hz, turn the rear speakers off, and measure. If a big dip appears, flip the sub's polarity. If there is a null (a deep dip) near 70 Hz, EQ won't help: move the sub, or set the crossover below that frequency and match the midbass.

4.2 #11 Polarity is not delay, and "half a wavelength of delay plus a polarity flip" re-aligns only one frequency.

> If the high-pass/low-pass phase relationship is wrong, delay fixes it only at the crossover point; that is the #1 reason bass won't come forward. A true half-wave path error at 80 Hz needs ~7 ft of path difference, so it is rare. RARE
>
> **Sources:** DIYMA 2016-01-22 · DIYMA 2021-08-19

In plain English: Polarity flips a wave upside down at every frequency. Delay shifts it in time, which shifts the phase by a different amount at each frequency. A wavelength is the length of one cycle of a sound wave; at 80 Hz it is about 14 ft (4.3 m). A common forum trick is to add half a wavelength of delay and then flip polarity. He says that re-aligns the speakers at only one frequency. If the crossover's high-pass and low-pass halves have the wrong phase relationship, delay can fix it only at the crossover point, and he calls that wrong relationship the #1 reason bass won't come forward. A real half-wavelength error caused by distance would need about 7 ft (about 2.1 m) of difference in path length at 80 Hz, so in a car it is rare.

4.2 #12 Set sub delay with a tape measure; it compensates distance, not crossover lag.

> At 80 Hz, 1 in of error is ~2° of phase, so being off by a lot hardly matters; the DSP can only add delay, so delay the nearer drivers to match the farthest. **Changed view:** 2009–2010 "don't time-align subs" (none needed with a crossover ≤80 Hz); from 2015 delay everything to the sub; 2018–2023 PDFs and 2021 posts tape-measure the sub too.
>
> **Sources:** DIYMA 2010-11-21 · DIYMA 2021-03-26 · V1 PDF 2018

In plain English: "Crossover lag" is the extra delay a crossover filter itself adds. He says to set the sub's delay with a tape measure, to make up for its distance only, not for crossover lag. At 80 Hz an error of 1 inch (2.5 cm) is only about 2° of phase, so even a sizeable measuring error hardly matters. A DSP can only add delay, never remove it, so you delay the nearer speakers to match the farthest one. The [Changed view] tag: in 2009–2010 he said not to time-align subs at all (none needed with a crossover at or below 80 Hz); from 2015 he said to delay everything to the sub; his 2018–2023 PDFs and 2021 posts tape-measure the sub along with the other speakers.

4.2 #13 Impulse-response "auto delay" is useless for subs.

> A low-passed channel's impulse peak moves with its high-frequency content, so the true arrival is where the trace first leaves zero (the zero crossing before the correct peak), and REW's own help says its sub delay estimate is inaccurate. The polarity-null delay test works only with even-order crossovers, matched levels and target slopes. RARE
>
> **Sources:** DIYMA 2016-03-25 · DIYMA 2021-09-02

In plain English: An impulse response is a measurement of how a speaker reacts to one sharp click; its graph shows a spike when the sound arrives. Some software, such as REW (a free measurement program), estimates delay automatically by finding that spike's peak. He says this is useless for subs. For a channel that plays only low notes, where the peak lands depends on how much high-frequency sound the channel has, so the peak is not the arrival time. The true arrival is where the trace first leaves zero, the zero crossing before the correct peak. He notes that REW's own help says its sub delay estimate is inaccurate. A common alternative is the "polarity-null" test: flip one speaker's polarity and adjust delay until the sound at the crossover cancels most deeply. He says it works only with even-order crossovers (such as LR2 or LR4), matched levels and slopes that match the target.

4.2 #14 Group delay is a "nothingburger" in car bass.

> It would matter only above ~50 ms, about one cycle at 20 Hz; people reject vented boxes over group delay, then add a 72 dB/oct sub low-pass that adds far more. A high-pass filter's group delay sits in its stop band, where the woofer isn't used. RARE
>
> **Sources:** DIYMA 2022-01-14 · DIYMA 2017-09-12

In plain English: Group delay is how much a filter or box delays one band of frequencies compared with others. Many people believe group delay makes vented boxes worse for sound quality. He calls it a "nothingburger" (a non-issue) in car bass. It would matter only above about 50 ms (fifty thousandths of a second), which is about one cycle of a 20 Hz wave. He points out that people reject vented boxes over group delay and then put a 72 dB/oct low-pass on the sub, which adds far more. And a high-pass filter's group delay sits in its "stop band", the range it cuts out, where the woofer isn't being used anyway.

4.2 #15 All-pass filters on sub and midbass: a last resort.

> An all-pass shifts phase over a band without changing level; a DSP "phase" slider (e.g. Helix) just moves an all-pass's center frequency. In 2015–16 a 2nd-order all-pass on the sub plus one half an octave higher on the near-side midbass centered bass in both seats of his car (aim ~90° or 270°, never 180°). RARE **Changed view:** 2010–2013 never made them work and called all-pass "front bass" tricks one-seat fixes;

> 2021 prefers fixing the crossover, and if used, puts it on the high-pass side, where that driver's level is falling.
>
> **Sources:** DIYMA 2013-03-16 · DIYMA 2015-10-19 · DIYMA 2021-09-26

In plain English: An all-pass filter shifts the phase (timing) of a band of frequencies without changing how loud they are. He says the "phase" slider on some DSPs, Helix for example, just moves an all-pass filter's center frequency. He calls all-pass filters on the sub and midbass a last resort. In 2015–16, in his own car, he used a 2nd-order all-pass on the sub plus another one, half an octave higher, on the nearer midbass. That centered the bass in both front seats. He aimed for a shift of about 90° or 270°, never 180°. The [Changed view] tag: in 2010–2013 he never made them work and called all-pass "front bass" tricks fixes for one seat only. In 2021 he preferred fixing the crossover itself. If an all-pass is used, he puts it on the high-pass side, where that speaker's level is already falling, so its effect on the combined sound is smaller.

4.2 #16 Each front seat sees a different sub-to-midbass phase, even with a center channel.

> It depends on sub location (he cites Harman patent US6876748); "pressure on the ears" can be a low-frequency phase problem, which he fixed in a CES demo car.
>
> **Sources:** DIYMA 2016-01-28 · DIYMA 2016-01-29

In plain English: A center channel is an extra speaker in the middle of the dash. Each front seat sits at different distances from the sub and from each midbass speaker, so the phase between sub and midbass is different in each seat. He says a center channel doesn't change that; it depends on where the sub sits. He cites a Harman patent on this, US6876748, about symmetrical stereo imaging in cars. He also says a feeling of "pressure on the ears" can come from a low-frequency phase problem; he fixed one in a demo car at CES (the big Consumer Electronics Show).

4.2 #17 Keep bass below ~100 Hz out of the rear speakers.

> His Logic 7 fix: rears high-passed at 250 Hz plus a separate 50–250 Hz band-pass with its own delay, level and EQ. Midbass arrays of 4–8 drivers evened bass across seats, but the more even, the worse the image; that is why BMW under-seat woofers and Mercedes "Front Bass" work. RARE
>
> **Sources:** DIYMA 2013-03-16 · DIYMA 2021-11-29

In plain English: He says rear speakers should get no bass below about 100 Hz. He adds one exception: when that rear bass is used to cancel some of the sub's sound in the rear seats, which are often much closer to the sub. Logic 7 is a Harman surround system that spreads stereo music across many speakers. His Logic 7 fix: the rear speakers get the ambience signal high-passed at 250 Hz, plus a separate 50–250 Hz band (a "band-pass", a slice between two frequencies) with its own delay, level and EQ. He also tried midbass "arrays" of 4, 6 and 8 drivers around the car. They evened out the bass across the seats, but the more even it got, the worse the image. He says that is why BMW's under-seat woofers and Mercedes' "Front Bass" work.

4.2 #18 Multichannel midbass: steer only common content to a center midbass.

> Below 50 Hz bass is nearly minimum-phase, so EQ fixes it; above that use left, center and right midbass, tune every location (sub included) to one target, then leave the sub gain alone.
>
> **Sources:** DIYMA 2009-11-14

In plain English: "Common content" is sound that is the same in the left and right channels, such as a centered bass guitar. He says only that shared part should be steered to a center midbass speaker. "Minimum-phase" is a technical term: in such a system, fixing the loudness with EQ fixes the timing too. He says car bass below 50 Hz is nearly minimum-phase, so EQ fixes it. Above that, use left, center and right midbass speakers, tune every location (the sub included) to one target curve, and then leave the sub's gain alone.

4.2 #19 The front-bass illusion has a ceiling.

> In his Volvo C30 demo car, tracks with bass mixed ~30 dB hotter than everything else still come from the back.
>
> **Sources:** OFCAP 38 @1:07:02

In plain English: Even done right, the illusion has limits. In his Volvo C30 demo car, tracks with bass mixed about 30 dB louder than everything else (a huge amount) still sound like the bass comes from the back. This fits his earlier point that bass far louder than the rest can't sound like it's in front (#1).

4.2 #20 Check bass polarity by ear and with long test signals; cheap pulse testers fail on subs.

> Fronts only, processing off, bass plus a centered vocal: a centered balance must give more bass than either end, or one midbass is reversed; never combine a polarity flip with delay. Pick the sub phase that gives the most midbass impact. Short (~9 ms) pulses miss subs (a 20 Hz cycle is 50 ms): use long positive pulses or a right-triangle wave; sub terminal markings are almost never wrong (3 of ~4,000). RARE
>
> **Sources:** DIYMA 2013-02-09 · Audiofrog blog Aug 2021 · DIYMA 2021-08-16

In plain English: Polarity is whether a speaker's cone moves outward when it should; reversed wires make it move backward. His checks, step by step. (1) By ear: play only the front speakers, with processing off, using a track with bass and a centered vocal. Slide the balance control from left to right. With the balance centered you should hear more bass than at either end; if not, one midbass is reversed. (2) Never combine a polarity flip with delay. (3) For the sub, pick the polarity that gives the most midbass impact. (4) Test signals: cheap pulse testers send short pulses, about 9 ms (nine thousandths of a second), but one cycle of a 20 Hz note lasts 50 ms, so they miss subs. Use a long series of positive pulses, or a right-triangle wave (one that jumps straight up and then ramps down). (5) The + and − markings on sub terminals are almost never wrong: in his count, 3 out of about 4,000.

4.3 Enclosures: vented vs sealed, sizing, ports

What this part is about: The box a subwoofer sits in. A sealed box is airtight; a vented (or ported) box has a tube or slot, the port, that lets air move in and out and adds bass of its own. He explains why he usually prefers vented boxes in cars, how to size boxes and ports, and which popular box ideas he thinks make little difference.

4.3 #1 He prefers vented boxes in cars, tuned for minimum cone excursion, with the in-car peak EQ'd out.

> Vented is more efficient across 30–50 Hz and, with a good port and one EQ band, has lower distortion and power demand and measures as flat as sealed. In a sealed box the sub must play below its in-box resonance (Fc), where the suspension distorts; "sloppy" vented bass only means a box tuned high for loudness. RARE
>
> **Sources:** DIYMA 2008-05-23 · DIYMA 2014-10-07 · DIYMA 2021-12-14

In plain English: A subwoofer ("sub") sits in either a sealed box or a vented box. Excursion is how far the cone moves back and forth; more excursion means more strain and more distortion (sound added that wasn't in the music). Many people believe sealed boxes are for sound quality and vented boxes are only for loudness. He disagrees. He prefers vented boxes in cars, tuned so the cone moves as little as possible, and he removes the bass bump the car then creates with EQ (an equalizer: a tone control that can cut or lift a narrow band of frequencies). His reasons: a vented box is more efficient across 30–50 Hz, and with a good port and one EQ band it has lower distortion, needs less power and measures as flat as a sealed box. In a sealed box, the sub has to play below Fc, the woofer's resonance in that box (the note where woofer and box naturally "bounce"), and there the suspension distorts. He says "sloppy" vented bass only comes from a box tuned high to be loud.

4.3 #2 Tune the port to the woofer and box, not to the car.

> Place the two impedance peaks at the passband edges so the port does most of the work, EQ the resulting peak, and check full-power excursion below Fb (the box tuning). Vented subs are almost always tuned below their sealed Fc; once a woofer is in a box there is no Fs, only Fc or Fb. RARE
>
> **Sources:** DIYMA 2011-10-05 · DIYMA 2021-03-05 · DIYMA 2021-05-09

In plain English: A vented box has a tuning frequency, Fb: the note at which the port resonates and does most of the work. Many people pick Fb to suit the car. He says tune the port to suit the woofer and the box instead. On an impedance graph (impedance is the electrical load the speaker presents to the amplifier), a vented box shows two peaks. He places them at the edges of the range the sub will play, with a wide trough between them, so the port does most of the work. In the car this gives a bass peak, which you cut with EQ. Then check an excursion plot at full power for frequencies below Fb, because below Fb the port stops holding the cone back. Fs is the bare woofer's resonance. He says that once the woofer is in a box there is no Fs any more, only Fc (sealed) or Fb (vented), and vented subs are almost always tuned below what their sealed Fc would be.

4.3 #3 A good vented box has less excursion everywhere above Fb and is not loudest at Fb.

> Choose volume and Fb so woofer plus port sum flat (they are ~90° apart at Fb: +3 dB, not +6); low tuning plus a boost near Fb is fine. Sealed's only edge: any sealed box for a given woofer sounds like any other. RARE
>
> **Sources:** DIYMA 2014-10-07 · DIYMA 2016-05-13

In plain English: Many people believe a vented box is built to be loudest at its tuning frequency (Fb). He says a good one is not, and that its cone moves less than it would in a sealed box at every frequency above Fb. You choose the box size and Fb so that the woofer's sound and the port's

sound add up to a flat response. At Fb the two are about 90° apart (a quarter of a cycle out of step). Two equal sounds exactly in step add 6 dB; a quarter cycle apart they add only about 3 dB (dB, decibels, measure loudness differences). Tuning low and adding an EQ boost near Fb is fine, he says. The one advantage he grants sealed boxes: any sealed box you build for a given woofer will sound much like any other sealed box for it.

4.3 #4 Keep the port and EQ the peak; but with no EQ at all, sealed is probably better.

> Turning the gain down to tame a vented peak leaves a hole at the crossover; compare boxes level-matched. Without EQ, sealed wins if you have the amp power and voice coil.
>
> **Sources:** DIYMA 2016-05-03 · DIYMA 2016-05-04

In plain English: In a car, a vented box makes a big bass peak (around 40 Hz in his example). Some people tame it by turning the sub's gain (its level) down until the peak matches everything else. He says that leaves a hole around the crossover, the point where the sub hands over to the door speakers. Better: keep the port open and cut just the peak with EQ. And when you compare a sealed and a vented box, match their levels first, or the comparison misleads you. His exception: with no EQ at all, a sealed box is probably the better choice, provided you have enough amplifier power and enough voice coil (the wire coil that drives the cone, which must cope with the heat).

4.3 #5 Put Fb near the bottom of the band and respect the power math.

> That minimizes excursion where it is worst and removes the need for a subsonic. 3 dB of boost doubles power, 12 dB is 16×, and each halving of frequency needs 4× the excursion.
>
> **Sources:** DIYMA 2008-12-18 · DIYMA 2008-12-05

In plain English: Fb is the vented box's tuning frequency. He says to put it near the bottom of the range the sub has to play. At Fb the port does the work and the cone barely moves, so this keeps excursion low exactly where it would otherwise be worst, and it removes the need for a subsonic filter (a filter that blocks notes too low for the box). Then he spells out the cost of bass. Every 3 dB of boost needs twice the amplifier power, and 12 dB needs 16 times the power. Each time the frequency halves (one octave lower), the cone must move 4 times as far to make the same loudness. So in practice: put Fb near the bottom of the band, shape the response with EQ, and remember how fast boosts use up power and cone travel.

4.3 #6 Small sealed box plus huge amp is "progress" in reverse, and a sub built for a big box fails in a tiny one.

> Typical small-box subs (~85 dB/W/m) need ~2,000 W to match an old Cerwin-Vega LE12 vented on a Rockford Punch 150. Squeezed into a tiny box, a sub that needs a big one makes no bass, and power plus EQ drive it past its limits. His 2023 C30 runs two GB10D4 in a small sealed box, per his rule "little space, plenty of power: sealed plus EQ". RARE
>
> **Sources:** DIYMA 2011-10-07 · DIYMA 2021-02-03 · OFCAP 38 @1:10:05

In plain English: Many people believe a small sealed box with a huge amplifier is the modern, better way to get bass. He calls it "progress" in reverse. Sensitivity says how loud a speaker plays from 1 watt, measured 1 meter away; about 85 dB/W/m is low. He says typical small-box subs need about 2,000 W to make the bass that an old Cerwin-Vega LE12 made in a vented box driven by a Rockford Punch 150 (an older amplifier). And a sub that needs a big box makes no bass when

squeezed into a tiny one; adding power and EQ to make up for it drives it past its limits. His own rule for little space and plenty of power is sealed plus EQ: his 2023 Volvo C30 demo car runs two Audiofrog GB10D4 subs in a small sealed box.

4.3 #7 Small sealed boxes don't give "more cone control", and box Q barely matters when the sub is crossed below Fc.

> A small box is a stiffer spring that raises resonance and Q. Crossed at 60 Hz, Qtc 0.707 vs 0.9 or 1 is immaterial (model the box with the low-pass applied); a too-small box shows up as weak lows and "one-note bass" near the crossover. RARE
>
> **Sources:** DIYMA 2012-07-28 · DIYMA 2021-02-03 · DIYMA 2022-01-13

In plain English: Many people believe a smaller sealed box gives "tighter" bass with "more cone control". He says it doesn't. The air in a small box acts like a stiffer spring, which raises the resonance and the Q. Q describes how much a system overshoots and rings at resonance (he contrasted a well-damped ripe avocado with a springy super ball). Qtc is the Q of a woofer in a sealed box. But when the sub is crossed over (filtered off) below Fc, as with his example of a 60 Hz crossover, whether Qtc is 0.707, 0.9 or 1 makes no real difference; to see this, model the box with the low-pass filter included. A box that is too small shows up as weak low bass and "one-note bass" near the crossover.

4.3 #8 Box type decides the Xmax you need.

> (Xmax: how far the cone moves before distortion becomes objectionable.) Sealed or IB at high power needs a long coil; vented can use a short one. Rules: Qts ≤0.4 with ≥10 mm linear Xmax suits sealed, vented woofers usually have Qts 0.4–0.7, ~7 mm "small-box" subs are usually better vented with a subsonic, and a 4.3 mm-Xmax sub "is a midrange". RARE
>
> **Sources:** DIYMA 2011-09-16 · DIYMA 2011-09-19 · DIYMA 2014-10-04

In plain English: Xmax is defined in the point itself. Qts is a number describing how well-controlled a woofer is at its resonance; lower means more controlled. He says the type of box decides how much Xmax you need. A sealed box or an infinite baffle (IB: the woofer on a board, with the trunk as its "box") at high power makes the cone travel far, so it needs a long voice coil. In a vented box the port does much of the work, so a short coil can do. His rules: Qts of 0.4 or less with at least 10 mm of linear Xmax suits sealed; vented woofers usually have Qts between 0.4 and 0.7; "small-box" subs with only about 7 mm of Xmax usually do better vented with a subsonic filter, even though they are marketed for small sealed boxes; and a sub with 4.3 mm of Xmax "is a midrange", not a subwoofer.

4.3 #9 Thiele/Small numbers are a box recipe, not a quality score, and "SQ sub" differences are mostly box response.

> Equal bass needs equal displacement (cone area × excursion). A bigger box only adds low bass: EQ it down and "tight" vs "boomy" disappear; heard differences without blind tests in one car come from elsewhere. Note Qts = (Qes × Qms)/(Qes + Qms). RARE
>
> **Sources:** DIYMA 2008-08-08 · DIYMA 2021-09-05 · DIYMA 2021-05-09

In plain English: Thiele/Small (T/S) parameters are the standard spec numbers of a woofer, such as Fs, Qts and Qes. Many people compare them as if they were a quality score. He says they are a recipe for choosing the right box, not a measure of how good a woofer is. To make the same bass, two subs must move the same amount of air (cone area times how far the cone travels). He says most of what people call "SQ sub" (sound-quality sub) differences is really the box's frequency response: a bigger box only adds low bass, and if you EQ that extra down, "tight" versus "boomy" disappears. Unless subs are compared blind in the same car, the differences people hear come from elsewhere in the system. The formula at the end is his correction of someone who had simply multiplied Qes and Qms.

4.3 #10 What sub distortion is, and why power compression hardly matters in a car.

> Odd-order "thud thud" comes mostly from the coil leaving the gap; even-order from asymmetry, above all cupped spiders. Subs alone run at and below resonance ("resonance IS distortion", though ~3% at Fs is fine); power compression only slightly steepens the low end of a sub low-passed at ~60 Hz. RARE
>
> **Sources:** DIYMA 2021-09-05 · DIYMA 2012-09-16

In plain English: Distortion is sound the speaker adds that wasn't in the signal. He sorts sub distortion into two kinds. Odd-order distortion, the "thud thud" sound, comes mostly from the voice coil moving out of the magnet gap, where the motor's force is. Even-order distortion comes from parts that aren't symmetrical, above all a cupped spider (the spider is the springy fabric ring that centers the cone). Subs are the only drivers that play at and below their resonance, and he says "resonance IS distortion", though about 3% at Fs is fine for a sub. Power compression is the loss of output as a voice coil heats up. He says it hardly matters in a car: for a sub low-passed at about 60 Hz it only makes the bottom end roll off slightly more steeply.

4.3 #11 Set the subsonic filter about half an octave below Fb (IB needs one too).

> Below Fb the box is "just a hole" and the cone unloads; on the A600.1D use 5 Hz when none is needed. Bass boost is costly: 15 dB at 45 Hz needs 32× the power.
>
> **Sources:** A-Series manual · DIYMA 2014-11-07

In plain English: A subsonic filter is a high-pass filter that keeps the very lowest notes away from the sub. Below Fb, he says, a vented box is "just a hole": the port no longer holds the cone back, so the cone "unloads" and can flap too far. Set the subsonic about half an octave below Fb (half an octave down means dividing the frequency by about 1.4). An infinite-baffle setup needs one too. The manual for Audiofrog's A600.1D amplifier says to set it to 5 Hz when no subsonic is needed. And bass boost is expensive: 15 dB of boost at 45 Hz needs 32 times the power.

4.3 #12 Size ports for air velocity at your power, with big flares; area rules of thumb are wrong.

> A choking port makes the box act sealed "with a nasty-sounding leak". More area needs more length; use big flares (a ¾" roundover barely helps), round beats slot, and a long 3" port on a 12" chokes, so go smaller woofer vented or seal and EQ. Audiofrog manuals give each sub a sealed box and a vented one (round flared or slot port). RARE
>
> **Sources:** DIYMA 2014-11-07 · DIYMA 2008-06-05 · Absolute Electronix 2023 @7:18

In plain English: The port is the tube or slot in a vented box. Many people size it with a rule of thumb based on port area and box volume. He says that's wrong: what matters is how fast the air moves through the port, which rises with the power you use (and depends on the cone size and motor). If the port is too small, friction "chokes" it at high power, and the box behaves like a sealed box "with a nasty-sounding leak". A port with more area must also be longer to keep the same tuning. Use big flares (rounded, trumpet-like ends); a ¾" (about 19 mm) roundover barely helps, and round ports beat slots. A long 3" (about 75 mm) port on a 12" sub chokes, so either use a smaller woofer in a vented box, or go sealed and use EQ. Audiofrog's manuals give each sub a sealed box and a vented one.

4.3 #13 Port position, port direction and baffle shape rarely matter; rigidity does.

> Port and cone would need ~16 ft of path difference to cancel at 35 Hz, so rear-firing ports are fine if the ends are clear. Brace the largest panels; a baffle would have to be ~5 ft across to matter at 100 Hz.
>
> **Sources:** DIYMA 2011-11-21 · DIYMA 2008-06-03

In plain English: Many people worry about where the port goes, which way it fires, or the shape of the front panel (the baffle). He says these rarely matter for subs. For the port's sound and the cone's sound to cancel at 35 Hz, their paths would have to differ by about 16 ft (about 5 m), so a port firing backward is fine as long as its ends aren't blocked. Likewise, a baffle would have to be about 5 ft (1.5 m) across to matter at 100 Hz. What does matter is stiffness: a panel that flexes makes its own sound, so brace the largest panels.

4.3 #14 Model with voice-coil inductance (Le), then EQ at playing level.

> A model without Le "isn't worth much" for multilayer car subs; build to a good model and you needn't measure the box. Large-signal shifts (a few Hz of Fb) are handled by EQ set at playing level; compare subs only after EQ'ing them to the same response.
>
> **Sources:** DIYMA 2008-12-06 · DIYMA 2015-05-29

In plain English: Box-design software predicts how a woofer will behave in a given box. He says the model must include voice-coil inductance (Le, an electrical property of the coil that affects how the woofer behaves as frequency rises). For car subs with multilayer coils, a model without Le "isn't worth much". If you build to a good model, you don't need to measure the box afterward. At real playing levels the woofer behaves a little differently (the tuning can shift by a few Hz); he handles that with EQ set at the level you actually listen at. And to compare two subs fairly, first EQ them to the same response.

4.3 #15 Polyfill lowers Qtc but absorbs nothing at sub frequencies; a pole vent may exit outside the box.

> Rear-wave reflections inside a sub box don't matter, and "deflex" pads add only panel mass. A 1–1.5" pole vent outside a sealed box changes little; route it into a small tube with inert material to keep noise out of the cabin.
>
> **Sources:** DIYMA 2015-11-15 · DIYMA 2022-01-04

In plain English: Polyfill is fluffy stuffing put inside sealed boxes. He says it lowers the box's Qtc but absorbs nothing at sub frequencies. Many people worry about sound bouncing around inside the box (the "rear wave"); he says those reflections don't matter in a sub box, and "deflex" pads only add a little mass to the panels. A pole vent is a hole through the middle of the woofer's magnet that lets air in and out behind the cone. He says a 1–1.5" (25–38 mm) pole vent that exits outside a sealed box changes little. To keep its noise out of the cabin, route it into a small tube with inert material.

4.3 #16 Aperiodic membranes and "impedance-peak removal" are wasted money, except in special cases.

> The resonance peak in impedance is just the driver's Q and is harmless. His one use: a membrane between an IB woofer and the listener to strip HF distortion. BMW under-seat boxes work as aperiodic boxes (frame airspace through small openings); high-Q drivers crammed in instead never sounded good to him. RARE
>
> **Sources:** DIYMA 2008-10-12 · DIYMA 2013-04-28 · DIYMA 2020-06-24

In plain English: An aperiodic membrane is a panel of cloth or foam that lets air leak slowly out of a box, damping the woofer's resonance. "Impedance-peak removal" means adding parts to flatten the bump in a speaker's electrical load at resonance. He says both are wasted money in almost every case: that peak simply reflects the driver's Q and does no harm. His one use for a membrane is between an infinite-baffle woofer and the listener, to strip away the woofer's high-frequency distortion. He adds that BMW's under-seat boxes work as aperiodic boxes, joining extra airspace in the car's frame through small openings. Setups that skip this and cram high-Q drivers in instead have never sounded good to him.

4.3 #17 A down-firing box is an accidental bandpass, not "boundary loading"; linked boxes are unpredictable.

> The space under the box and the gap out act as a front chamber that is nearly impossible to model (carpet losses, no clear port). Passive radiators are fine, but two boxes joined by an internal port shift tuning with power: use two sealed boxes plus EQ. RARE
>
> **Sources:** DIYMA 2021-10-29 · DIYMA 2013-05-21 · DIYMA 2013-04-28

In plain English: A down-firing box has its woofer facing the floor. Many people say this gives "boundary loading" (extra bass from sitting near a surface). He says it really makes an accidental bandpass box. A bandpass box is one where the woofer fires into a chamber and the sound leaves through an opening, which acts as a low-pass filter. Here, the space under the box and the gap the sound escapes through form that front chamber, and it is nearly impossible to model, because carpet absorbs energy and there is no clear port. Passive radiators (an unpowered cone used in place of a port) are fine. But two boxes joined by an internal port change their tuning as power changes; he says use two sealed boxes plus EQ instead.

4.3 #18 Sub "break-in" is spider resin cracking.

> It lowers Fs and Qts (sealed: Fc and Qtc drop; vented: more output above Fb); mids and tweeters crossed above resonance barely break in.
>
> **Sources:** DIYMA 2010-05-09 · DIYMA 2021-05-07

In plain English: "Break-in" is the idea that a new speaker changes as it is played in. He says that for subs it is real and has a simple cause: the resin that stiffens the spider (the springy ring that centers the cone) cracks with use. That lowers Fs and Qts. In a sealed box, Fc and Qtc drop; in a vented box you get more output above Fb. Midranges and tweeters, crossed over above their resonance, barely break in.

4.3 #19 The manual box is the right box, not a "safe" one: Audiofrog designs woofers backwards from the car.

> It starts with an in-car target averaged over ~30 cars, picks a convenient box, then designs the woofer for that box (modeled in LEAP); midband sensitivity specs don't predict in-car sub loudness. GB10: 17.6 mm Xmax plus ~8 mm more travel, so keep grilles ~1.25" from the cone. RARE
>
> **Sources:** Absolute Electronix 2023 @8:13 · DIYMA 2021-09-04 · DIYMA 2021-12-22

In plain English: Many people treat the box in a sub's manual as a cautious "safe" choice and build something else for more bass. He says the manual box is the right box, because Audiofrog (his company) designs its woofers backward from the car. It starts with an in-car target response averaged over about 30 cars, picks a convenient box, and then designs the woofer for that box, modeled in the LEAP software. He adds that the sensitivity figure in a spec sheet (measured in the midband) doesn't predict how loud a sub will be in a car. For the GB10, the cone can travel 17.6 mm (its Xmax) plus about 8 mm more, so keep grilles about 1.25" (32 mm) from the cone.

4.3 #20 Audiofrog subs (product-specific).

> GB10/GB12 are "dual impedance" (GB10D4: 8 or 2 Ω; D2: 4 or 1 Ω) with 3" copper coils; a GB D4 at 2 Ω on an A600.1D is "the right amount of power". GS8ND2: under 3" deep, 10 mm Xmax, sealed ~7 L, vented ~14 L or IB, plays to ~300 Hz, ~2 mm excursion at rated power crossed at 80 Hz.
>
> **Sources:** CMA 2021 @27:03 · GS8ND2 store listing · DIYMA 2020-06-24

In plain English: This point is about Audiofrog's own subs. GB10 and GB12 are "dual impedance": each has two voice coils that can be wired for either of two loads (the GB10D4 as 8 or 2 ohms, the D2 version as 4 or 1 ohm; ohms measure the electrical load). They use 3" copper voice coils. He says a GB D4 wired to 2 ohms on Audiofrog's A600.1D amplifier is "the right amount of power". The GS8ND2 is a very shallow woofer (under 3" deep) with 10 mm of Xmax. It works sealed in about 7 liters, vented in about 14 liters or as an infinite baffle, plays up to about 300 Hz, and moves only about 2 mm at its rated power when crossed at 80 Hz.

4.3 #21 His earlier designs, and an Xmax method conflict.

> JBL GTi subs used opposed dual coils (Differential Drive); the 2012 MS-BassProSQ used a short coil and a flared port tuned for minimum excursion; he designed the Infinity Perfect 10 VQ (variable-Q inserts). Compare Xmax only within one method. **Conflict:** 2008, JBL Xmax was one-way overhang "with no +10% fudge"; 2020, GB60's 9 mm was (coil − gap)/2 + 15%, vs Klippel ~4.7 mm.
>
> **Sources:** DIYMA 2008-12-26 · Erin's Audio Corner Aug 2020 · DIYMA 2021-12-15

In plain English: This point covers subs he designed before Audiofrog, and a conflict about how Xmax is measured. At JBL, the GTi subs used two opposed voice coils (Differential Drive). The 2012 MS-BassProSQ used a short coil and a flared port tuned for minimum excursion. He also designed the Infinity Perfect 10 VQ, which came with inserts that change its Q ("variable Q"). Xmax can be

calculated in different ways. The conflict, in simple words: in 2008 he described JBL's Xmax as the coil's one-way overhang "with no +10% fudge", that is, no added margin. In 2020, the GB60's 9 mm figure was worked out as (coil height minus gap height) divided by 2, plus 15%, while a Klippel measurement (a lab system that measures a driver's actual usable travel) gave about 4.7 mm. The point takes no side; its advice is simply to compare Xmax figures only when they were worked out the same way.

4.4 Infinite baffle (IB: woofer on a baffle, trunk as the "box")

What this part is about: Infinite baffle (IB) means mounting the woofer on a sealed board, usually the rear shelf, so the trunk serves as a very large box. This part covers how he installs IB subs, how he tunes them, which woofers suit them, and what happens when bass fires through the rear seat.

4.4 #1 Fire IB woofers through the package tray, rebuild only the deck, and don't seal the seat back or trunk.

> Seal the tray openings and kill rattles (MDF baffle, deadener, foam between trim and deck); seal the seat only if it's thin and unbacked. Never seal the trunk (cars vent cabin air through it); his own IB sub hangs on rubber isolation mounts. RARE
>
> **Sources:** DIYMA 2009-08-24 · DIYMA 2014-07-20 · DIYMA 2017-12-09

In plain English: The package tray is the shelf behind the rear seat, under the back window. In an IB install the woofers are mounted there and fire up into the cabin. Many people believe IB needs the whole trunk and seat back sealed airtight. He says rebuild only the deck: seal the openings in the tray and kill rattles, using an MDF board as the baffle, deadener (heavy sticky damping sheet) and foam between the trim and the deck. Seal the seat back only if it is thin and has nothing behind it. Never seal the trunk, because cars vent cabin air out through it. His own IB sub hangs on rubber isolation mounts, which he used to avoid rattles.

4.4 #2 EQ the cabin gain out instead of re-engineering the trunk.

> Trunk volume changes ~6 dB at 20 Hz while the car adds ~20 dB: cut much of the cabin gain, flatten to 150–200 Hz, low-pass to suit the midbass, set level with gain — faster than "a thousand trips to Home Depot". Dacron-lined trunks and holes to the outside are "ridiculous". RARE
>
> **Sources:** DIYMA 2013-05-11 · DIYMA 2013-05-05

In plain English: Cabin gain is the car's natural boost of low bass. Many IB installers spend a lot of effort modeling and changing the trunk as if it were a precise box. He says don't bother: changing the trunk's volume moves the bass by about 6 dB at 20 Hz, while the car itself adds about 20 dB. So use EQ instead: cut much of the cabin gain, flatten the response up to 150–200 Hz, set the low-pass to suit your midbass, and set the level with the amplifier's gain. He says that is faster than "a thousand trips to Home Depot". He calls lining the trunk with Dacron (polyester batting) or cutting holes to the outside "ridiculous".

4.4 #3 IB woofers need Qts ≤~0.7, enough Xmax for the power, and a subsonic.

> In a car, sealed plus cabin gain is about flat, vented peaks above Fb, and IB rises as frequency falls; IB handles less power but needs less. Low-Qts (~0.38) "small sealed" woofers in IB act like a vented box: cut the peak above ~36 Hz and raise the gain.
>
> **Sources:** DIYMA 2009-08-24 · DIYMA 2018-02-02 · DIYMA 2014-07-25

In plain English: His requirements for an IB woofer: a Qts of about 0.7 or less, enough Xmax for the power you will use, and a subsonic filter. He compares the three setups in a car: a sealed box plus cabin gain comes out about flat; a vented box peaks above its tuning (Fb); and IB rises steadily as frequency falls. IB handles less power, but it also needs less. Woofers sold for "small sealed" boxes with a low Qts (around 0.38) act like a vented box when used IB, so cut the peak above about 36 Hz with EQ and raise the amplifier gain.

4.4 #4 IB in a BMW 3-series: cut the tray and use metal or plastic mesh, never grille cloth.

> IB is "the easiest installation in the world" and removes the "peaky sealed box sound"; cloth can snap against the mesh.
>
> **Sources:** DIYMA 2012-09-03

In plain English: This is about BMW 3-series cars, where the sub can fire through the rear shelf. He calls IB "the easiest installation in the world" and says it removes the "peaky sealed box sound". Cut the tray open and cover the hole with metal or plastic mesh, never grille cloth, because the cloth can snap against the mesh as the air moves.

4.4 #5 Firing through the rear seat is an acoustic low-pass filter.

> It removes midbass, so add strong midbass up front; an 8" behind a 5" opening only low-passes far above an 80 Hz crossover.
>
> **Sources:** DIYMA 2009-08-24 · DIYMA 2012-08-12

In plain English: Some trunk or IB subs play into the car through the rear seat. He says the seat acts as an acoustic low-pass filter: deep bass gets through but midbass (the upper bass) is lost, so you need strong midbass speakers up front. A small opening is different: an 8" woofer behind a 5" hole is also a low-pass, but only at frequencies far above an 80 Hz crossover, so it doesn't matter (as long as it isn't noisy).

4.5 Sub location, number of subs, multiple-sub wiring

What this part is about: Where to put the subwoofer, how many to use and how to wire more than one. It also covers the "cross-car mode": a dip in the bass, usually near 70 Hz, caused by sound bouncing across the width of the car.

4.5 #1 Trunk subs work; trunk midbass doesn't.

> Sub bass reaches the cabin (you hear the next car's at a light), but midbass may not, and a small hole won't help: use a bandpass box firing through the opening, or real front midbass.
>
> **Sources:** DIYMA 2012-01-22 · DIYMA 2013-05-03

In plain English: Subs in the trunk work: low bass gets into the cabin, which is why you can hear the bass of the car next to you at a traffic light. Midbass (the upper bass, just above what a sub plays) may not get through, and cutting a small hole between trunk and cabin won't help much. If you want midbass from the trunk, use a bandpass box whose port fires through the opening; otherwise use real midbass speakers up front.

4.5 #2 Behind the rear seat (mid-car) is the worst sub location, and a deep null at the seat there is placement, not crossover phase.

> It is the easiest spot to design for, but his experience and a member's measurements rank it last. RARE
>
> **Sources:** DIYMA 2021-09-16

In plain English: Behind the rear seat, roughly the middle of the car, is the easiest spot to design a sub box for. He says it is the worst sub location, based on his experience and a forum member's measurements. A null is a deep, narrow drop in the response. He says a deep null at the seat with the sub there comes from where the sub and the listener sit, not from a phase problem (the sub and front speakers pushing and pulling out of step) at the crossover.

4.5 #3 Confirm the cross-car mode with a tape measure.

> Measure sub → mic, then sub → far side of the car → mic; a difference near 7 ft is the standard cross-car mode (the dip near 70 Hz). Move the mic to the other seat to see whether the dip follows; one dual-sub dip vanished when the subs were crossed at ~60 Hz. RARE
>
> **Sources:** DIYMA 2021-12-02 · DIYMA 2021-12-05

In plain English: A mode is a frequency at which sound bouncing between surfaces reinforces or cancels itself. The standard one across the car's width shows up as a dip near 70 Hz. His tape-measure test: measure the distance from the sub straight to the microphone, then from the sub to the far side of the car and back to the mic. If the difference is close to 7 ft (about 2.1 m), the dip is most likely the cross-car mode. Then move the mic to the other front seat and see whether the dip follows. In one member's dual-sub box, the dip disappeared once the subs were crossed over at about 60 Hz.

4.5 #4 Under-seat woofers in the BMW layout are "optimum".

> Center-of-car woofers don't blast rear passengers (who usually get bass 6–9 dB too hot), the seat hides their distortion, and left/center/right steering works over their range. In his MS-8 BMW build they played ~80–200 Hz (200 Hz because the 4" doors couldn't go lower loudly), with the sub below ~60 Hz; keep an octave of usable bandwidth around each crossover. RARE
>
> **Sources:** DIYMA 2013-03-10 · DIYMA 2021-08-04

In plain English: Many late-model BMWs have woofers under the front seats, and he calls this layout "optimum". Because the woofers sit in the middle of the car, they don't blast the rear passengers (who usually get bass 6–9 dB too loud), the seat hides their distortion, and a processor can steer sound left, center or right across their range. In a BMW he set up with the MS-8 (a JBL processor from his Harman years), the under-seat woofers played about 80–200 Hz. The top was 200 Hz because the 4" door speakers couldn't play any lower loudly; the sub covered about 60 Hz and below. His rule: keep about an octave of a speaker's usable range around each crossover.

4.5 #5 Multiple subs: one amp, parallel wiring, and EQ, not mode-smoothing arrays.

> Wire separate woofers in parallel (one DVC woofer's coils in series is fine); at very low frequencies two subs EQ'd behave like one. Mixed impedances split power unevenly: in parallel the lower impedance takes more (~67% for a 2 Ω DVC beside a 4 Ω coil), in series the higher. Mode-smoothing arrays suit subs only; above that they break the image. RARE
>
> **Sources:** DIYMA 2012-03-10 · DIYMA 2022-02-11 · DIYMA 2015-06-07

In plain English: For more than one sub, he uses one amplifier, parallel wiring and EQ. Wire separate woofers in parallel; inside one dual-voice-coil (DVC) woofer, wiring its two coils in series is fine. At very low frequencies, two subs that have been EQ'd behave like one. If you mix different impedances, power splits unevenly: in parallel the lower-impedance load takes more (about 67% for a 2-ohm DVC woofer next to a 4-ohm coil), and in series the higher-impedance one takes more. Mode-smoothing arrays are groups of woofers placed to even out bass across seats. He says they suit subs only; above the sub range they break the image (the sense that the music sits on a stage in front of you).

4.5 #6 Not enough bass? His four options, in order: vented instead of sealed, lower impedance, more speakers, a more efficient driver.

> Lower impedance adds little because amps don't double into half the load; "bottoming" may be amp clipping, and either way you need "more power or more woofer". For a sub amp, take the most power on offer (~800 W in one case) and bridge a 2-channel amp for single-coil GB woofers.
>
> **Sources:** DIYMA 2010-03-16 · DIYMA 2021-07-10

In plain English: If a system lacks bass, he lists four fixes, in order: change a sealed box to vented, lower the impedance, add more speakers, or use a more efficient driver. Lowering impedance adds little, because amplifiers don't double their power into half the load. What sounds like the sub "bottoming" (hitting the end of its travel) may really be the amplifier clipping (running out of voltage and flattening the tops of the waveform). Either way, he says, you need "more power or more woofer". For a sub amp, take the most power on offer (about 800 W in one case he advised on). Audiofrog GB woofers have a single voice coil, so a 2-channel amp should be bridged (its two channels combined into one) to drive one.

4.5 #7 Blown subs: suspect reversed DVC coil wiring and clipping, not DC.

> Out-of-phase coils make little bass, so owners crank gain and boost into clipping; it takes ~28 V DC at 1 Ω to make 750 W, and no loud pop at amp shutdown means no meaningful DC.
>
> **Sources:** DIYMA 2011-09-16

In plain English: When subs keep blowing, many people blame DC (direct current) leaking from the amplifier. He says to suspect reversed coil wiring and clipping instead. Picture a dual-voice-coil sub with one coil wired backward: the coils fight each other, so it makes little bass, and owners crank the gain and bass boost until the amp clips. A clipped signal carries more average power, which overheats the coil. DC is unlikely: making 750 W into 1 ohm would take about 28 volts of DC. If you don't hear a loud pop when the amp shuts off, there is no meaningful DC.

4.6 Midbass capability and the midbass/sub split

What this part is about: Midbass is the upper bass just above the subwoofer's range, usually played by woofers in the doors or kick panels (the panels in the footwells). This part covers how low midbass speakers should play, where to split the bass between them and the sub, and where to mount them.

4.6 #1 Crossing door midbass at ~40 Hz to pull the bass forward is "the worst mistake".

> Door midbass Fs is typically 60–70 Hz; below resonance the spider does the work, distorts most and wears out, and every car's cross-car mode near 70 Hz needs another fix. When told mids are crossed at 40 Hz he "stops listening"; making doors play 50 Hz is "the worst possible advice". RARE
>
> **Sources:** OFCAP 19 @1:45:06 · DIYMA 2018-02-09 · DIYMA 2022-01-14

In plain English: Many people make door midbass play down to 40–50 Hz so that the bass seems to come from the front. He calls crossing door midbass at about 40 Hz "the worst mistake". Door midbass drivers usually have their resonance (Fs) at 60–70 Hz. Below resonance, the spider (the springy ring that centers the cone) does the work; that is where the driver distorts most, and the spider wears out. Every car also has a cross-car mode (a cancellation across the car's width) near 70 Hz, which then needs a separate fix. When someone says their mids are crossed at 40 Hz, he "stops listening", and he calls making doors play 50 Hz "the worst possible advice".

4.6 #2 Cross every driver except the sub at or above its resonance: about 70–80 Hz for a 6" midbass.

> "Resonance IS distortion"; 80 Hz raises output, lowers distortion and lengthens suspension life. Each octave lower costs ~2× power and 4× excursion, so don't run a small speaker (e.g. a GB40) down to 100 Hz either. RARE
>
> **Sources:** DIYMA 2017-08-17 · DIYMA 2021-09-11 · DIYMA 2021-12-03

In plain English: His rule: cross over every speaker except the sub at or above its resonance frequency. For a typical 6" midbass that means about 70–80 Hz. He says "resonance IS distortion", so crossing at 80 Hz raises output, lowers distortion and makes the suspension last longer. Each octave lower (each halving of frequency) costs about twice the power and four times the cone travel. The same thinking applies to small speakers: don't run one such as the GB40 (a small Audiofrog speaker) down to 100 Hz either.

4.6 #3 A door or kick is a corner worth ~6 dB, so car midbass should be low-Q.

> Each halving of radiating space adds ~6 dB below ~500 Hz; home 6" drivers fake bass with high Q ("That higher Q IS distortion"), and the GB60 sounds thin on a display board but full in a car. RARE
> [Changed view (partial): 2021, crossed above resonance a midbass's Q "is not really important".]
>
> **Sources:** Audiofrog blog Jun 2021 · DIYMA 2015-12-13 · DIYMA 2021-09-11

In plain English: Below a few hundred hertz, a speaker sends sound in all directions, so nearby surfaces reflect its sound and add to it. He says each halving of the space a speaker radiates into adds about 6 dB below about 500 Hz, and a door bottom or kick panel works like a corner, worth about 6 dB. So car midbass speakers should be low-Q (well controlled at resonance). Home 6" speakers often fake bass with a high Q, and he says "That higher Q IS distortion". That is why the

Audiofrog GB60 sounds thin on a store display board but full in a car. The tag marks a partial change of view: in a 2021 forum post he said that once a midbass is crossed over above its resonance, its Q "is not really important". So the two positions are that car midbass should be low-Q, and that Q matters little when crossed above resonance.

4.6 #4 Ultra-low-Fs (~35 Hz) midbass drivers chase a crossover you don't need once there's a sub.

> One was nearly 10 dB less sensitive in midbass than midband ("10dB is a lot of power"), and "50 Hz isn't midbass". A sub isn't a good midbass either: midbass drivers use compliant suspensions and ~¼ the inductance for efficiency (Hoffman's Iron Law), so a "sub that plays midrange" mostly doesn't exist. RARE
>
> **Sources:** DIYMA 2021-11-02 · DIYMA 2022-01-13 · DIYMA 2020-08-24

In plain English: Some midbass drivers are built with a very low resonance (Fs around 35 Hz) so they can be crossed very low. He says that chases a crossover you don't need once you have a sub. One such driver was nearly 10 dB less sensitive in the midbass than in its midband, and he notes "10dB is a lot of power" (10 dB takes ten times the power). He also says "50 Hz isn't midbass". The reverse doesn't work either: a sub is not a good midbass. Midbass drivers use more compliant (softer) suspensions and about a quarter of a sub's coil inductance to be efficient. Hoffman's Iron Law is the standard rule that a speaker can't be small, efficient and deep-playing all at once. So, he says, a "sub that plays midrange" mostly doesn't exist.

4.6 #5 With a sub and an active high-pass, door enclosures are unnecessary, and never port a door box into the door.

> The high-pass does what a box would; damp the door and mount free-air. Vent into the door and you hear only the woofer, less bass than sealed. GB60 band ~70 Hz–3.5 kHz; in a GB four-way, midbass high-pass 60–80 Hz and low-pass ~300 Hz. RARE
>
> **Sources:** DIYMA 2016-06-17 · DIYMA 2022-01-13 · CMA 2022 @57:02

In plain English: Many people build boxes inside doors for their midbass. He says that with a sub and an active high-pass filter (a processor or amp setting that keeps deep bass away from the midbass), door enclosures are unnecessary, because the high-pass does what a box would do. Damp the door (deaden it) and mount the speaker "free-air", with no box. And never port a door box into the door: you then hear only the woofer, losing what makes a ported box worthwhile, and get less bass than a sealed box would give. Audiofrog's band for the GB60 is about 70 Hz–3.5 kHz; in a GB four-way system, the midbass high-pass is 60–80 Hz and its low-pass about 300 Hz.

4.6 #6 A 60–80 Hz dip in only the near-side midbass is likely the cross-car mode: cross above it and let the sub play there.

> Near-field measurements can't show door-panel vibration; near field turns far field at ~3× the driver's largest dimension. RARE
>
> **Sources:** DIYMA 2022-03-16

In plain English: If only the midbass on the side nearer you shows a dip between 60 and 80 Hz, he says it is likely the cross-car mode (the cancellation across the car's width). His fix: cross the midbass over above that dip and let the sub play there. He adds two measuring notes. Measuring

very close to the speaker (the near field) can't show the door panel vibrating. And the near field turns into the far field at about three times the driver's largest dimension.

4.6 #7 The passenger kick panel measured as the strongest midbass spot; the door's top corner evens left/right.

> His old Jetta tests matched a member's; with a sub, no 8" is needed.
> **Sources:** DIYMA 2021-09-16 · DIYMA 2021-12-03

In plain English: His old measurements in a Jetta, like a forum member's, found the passenger kick panel (the panel in the footwell) to be the strongest midbass location. Mounting the midbass in the top corner of the door makes the left and right responses much more alike. And if you have a sub, he says you don't need an 8" midbass.

4.6 #8 At moderate levels a 4–5" can blend with a sub; dedicated stereo midbass matters at real volume.

> It needs separate L/R parametric EQ; modern mixes carry stereo bass (mono bass was a vinyl limit).
> **Sources:** DIYMA 2008-11-20 · DIYMA 2021-08-20

In plain English: At moderate volume, even a small 4–5" speaker can blend with a sub, given separate left and right parametric EQ (an EQ where you choose each band's frequency, width and level). At real volume, dedicated midbass speakers in stereo (separate left and right) matter. Many people believe bass is mono, so stereo midbass wouldn't matter. He says modern recordings carry stereo bass; bass was mono on vinyl records only because of the medium's limits.

4.6 #9 Most cars have too much 60–200 Hz; thick midbass is his pet peeve.

> MS-8 usually turns that band down "because all cars have way too much of that".
> **Sources:** DIYMA 2012-03 · OFCAP 19 @3:31:01

In plain English: He says most cars have too much sound between 60 and 200 Hz, and thick, heavy midbass is his pet peeve. The MS-8 (a JBL processor from his Harman years) usually turned that band down, "because all cars have way too much of that".

4.6 #10 Midbass power needs are modest (product-specific).

> At 115 W with an 80 Hz crossover a GB60 never reaches 3 mm of excursion. Without a sub, a GS690 6x9 makes more bass; with a sub, the GB60 (2" coil, more than twice the Xmax).
> **Sources:** DIYMA 2021-07-21 · DIYMA 2021-08-08

In plain English: This point is about specific Audiofrog products. Midbass speakers need modest power: at 115 W with an 80 Hz crossover, the GB60 never moves as much as 3 mm. If you have no sub, a GS690 (a 6x9" oval speaker) makes more bass; with a sub, he picks the GB60, which has a 2" voice coil and more than twice the Xmax.

4.7 How much power you really need, and what clipping does

What this part is about: How many watts each speaker in a car really uses, why only the subwoofer amplifier ever works near full power, and how "clipping" (an overdriven signal) is what actually burns speakers.

4.7 #1 Power demand halves every octave up, so only the sub amp ever reaches rated power.

> Pink noise (equal energy per octave) mimics music: calibrate 20 W at 20 Hz and a tweeter crossed at 2.5 kHz gets ~0.16 W (~0.02 W at 20 kHz), and a midbass crossed at 80 Hz needs at most a quarter of the sub's power. A tweeter needs "a watt or two at the most". RARE
>
> **Sources:** Audiofrog blog Dec 2020 · DIYMA 2020-10-14 · DIYMA 2020-12-23

In plain English: Sound is measured in hertz (Hz), vibrations per second: deep bass is around 20–80 Hz, treble is thousands of Hz. An "octave" is a doubling of frequency, such as 20 Hz to 40 Hz. Pink noise is a test hiss with equal energy in every octave, and he uses it as a stand-in for real music. In pink noise, the power at any one frequency halves each time the frequency doubles. So if an amp makes 20 W at 20 Hz, it makes 10 W at 40 Hz, 5 W at 80 Hz, and so on, down to about 0.16 W at 2.5 kHz and about 0.02 W at 20 kHz. A tweeter (the small treble speaker) that starts playing at 2.5 kHz therefore gets a tiny fraction of a watt. A midbass speaker that starts at 80 Hz (two octaves above 20 Hz) needs at most a quarter of what the sub gets. Many people believe tweeters need big amplifiers. His point is that only the sub amp ever reaches its rated power, and a tweeter needs "a watt or two at the most".

4.7 #2 Measured: real music averages about 1/32 of its peak power, most of it bass.

> An EDM track peaking near 0 dBFS (call it 400 W) averaged ~−16 dB (he rounds to −15 dB = 1/32 of the power, ~12.5 W), with peak energy near 50 Hz and 1 kHz about 20 dB down.
>
> **Sources:** DIYMA 2021-11-29

In plain English: In a digital recording, "0 dBFS" is the loudest level the format can hold. A decibel (dB) is a ratio scale for loudness or power: every 10 dB down is one tenth of the power. He measured an electronic dance music (EDM) track whose loudest moments hit about 0 dBFS. If those peaks drove an amp to 400 W, the track's average was about 16 dB lower, which he puts at about 12.5 W, roughly 1/32 of the peak. Most of that energy sat in the bass, near 50 Hz. At 1 kHz (the middle of the audio range) the level was about 20 dB lower, which is one hundredth of the power. The question he was answering was about an amp's average current draw. His point is that even loud music uses only a small slice of the amp's peak power on average, and most of what it does use is bass.

4.7 #3 Rule of thumb: 50–100 W for mids and highs, at least ~800 W on the sub.

> Cars need 6–12 dB more bass to sound natural (~9 dB takes ~8× the power), while extra watts elsewhere buy little: 80–125 W is 1.9 dB, and a 90 dB/1 W speaker turns 99.4% of it into heat. A speaker 3 dB more sensitive equals doubling the amp.
>
> **Sources:** DIYMA 2013-04-25 · DIYMA 2017-07-05 · DIYMA 2022-01-26

In plain English: His rough guide is 50–100 watts per channel for the midrange and treble speakers, and at least about 800 W for the subwoofer. The reason is that a car needs 6–12 dB more bass than the rest of the range to sound natural. Every 3 dB more takes double the power, so about 9 dB more takes about 8 times the power. Extra watts on the other speakers buy little: going from 80 W to 125 W is only 1.9 dB louder, a small change. A speaker's "sensitivity" is how loud it plays from 1 watt (here 90 dB). He notes that such a speaker turns 99.4% of the power into heat, not sound. So a speaker that is 3 dB more sensitive gives the same gain as doubling the amplifier's power.

4.7 #4 You can't burn a coil with too little power; the extra high-frequency power of a clipped wave does it.

> A clipped wave approaches a square, whose average power equals its peak (a sine's is half), so a 100 W amp clipped hard makes ~200 W, all the extra at high frequencies; clipping is not DC. Subs survive it (coil inductance and class-D output filters reject the HF); tweeters behind a passive crossover take it. RARE
>
> **Sources:** OFCAP 19 @20:07 · DIYMA 2021-07-01 · DIYMA 2021-07-08

In plain English: Many people believe a small, "underpowered" amp blows speakers. He says too little power can't burn a speaker's voice coil (the wire coil that drives the cone). What does damage is clipping. Clipping happens when an amp is asked for more than it can give, so the smooth tops of the sound wave get cut flat. A pure tone is a smooth "sine" wave; a hard-clipped one looks almost like a flat-topped "square" wave. A square wave's average power equals its peak, while a sine's average is half its peak. So a 100 W amp clipped hard puts out about 200 W, and all the extra is high-frequency content. He also says clipping is not DC (a steady one-direction current), another common belief. Subwoofers survive this because their coils naturally block high frequencies, and class-D amps have output filters that remove some of it. Tweeters fed through a passive crossover (a filter box of coils and capacitors at the speakers) receive that extra treble energy and burn.

4.7 #5 Too much amp kills tweeters even without clipping; "overpowering for safety" works only if nothing can clip.

> A 200 W amp at full output feeds a "20 W" speaker's tweeter ~2 W, ten times its share. Gains set on a −10 dB track license a 300 W amp to try for 3,000 W, and a 250 W/ch amp bridged onto a 100 W speaker melts the coil. RARE
>
> **Sources:** DIYMA 2020-10-14 · OFCAP 19 @27:00 · OFCAP 19 @36:04

In plain English: Many people believe you should buy an amp bigger than the speaker's rating "for safety". He says that only works if the amp can never clip. Using his pink-noise math, the tweeter in a speaker rated "20 W" handles about 0.2 W. A 200 W amp at full output gives that tweeter about 2 W, ten times its share, with no clipping at all. The "gain" is the amp's input-level knob. If you set it using a test track recorded 10 dB below maximum, a full-level song asks for 10 times the power (10 dB is 10× power). A 300 W amp is then trying to make 3,000 W, so it clips. In the same talk he says a 300 W amp on a 100 W speaker is safe only if the gain is set with a full-level (0 dB) test track. His last example: a 250 W per channel amp "bridged" (two channels combined into one bigger one) onto a speaker rated 100 W melts the voice coil.

4.7 #6 Head-unit or DSP clipping, not amplifier clipping, usually kills mids and tweeters.

> About 70% of GB25 warranty claims are DSP input clipping (25% are broken terminal screws), and returned "crackling" units measure perfect. Clipping the A/D shows up in the mids and tweeters, not the sub, and "sounds like broken speakers". RARE
>
> **Sources:** Audiofrog blog Mar 2021 · DIYMA 2022-03-13 · Absolute Electronix 2023 @33:35

In plain English: The head unit is the radio in the dash. A DSP (digital signal processor) is a box that splits, delays and equalizes the signal before the amps. Its first stage, the A/D (analog-to-digital) converter, turns the incoming voltage into numbers, and if the voltage is too high it clips. He says this is usually what damages midranges and tweeters, not the amplifier clipping. His evidence: about 70% of warranty claims on Audiofrog's GB25 midrange are really DSP input clipping, and 25% are broken terminal screws. Speakers sent back for "crackling" measure perfect. This kind of clipping shows up in the mids and tweeters, not the sub, and "sounds like broken speakers". So a crackle may be a setup error, not a bad speaker.

4.7 #7 Find where distortion lives by turning things down, starting at the source.

> If it stays with the head unit turned down, it isn't after the head unit; if it stays with the phone turned down, it's the track or codec (streaming mixes are often clipped, and phone "volume leveler" features add more). Ask what changed (new phone, new service?); no bass over Bluetooth means a fault, since codecs strip only highs. RARE
>
> **Sources:** DIYMA 2021-10-13 · DIYMA 2022-01-15 · DIYMA 2021-07-08

In plain English: This is a way to track down where a distortion comes from by turning down one part of the chain at a time, starting at the source. If the distortion is still there with the head unit turned down, it isn't coming from anything after the head unit. If it is still there with the phone turned down, it is in the track itself or the codec. A codec is the compression method used to stream or send audio, for example over Bluetooth. He notes that streaming mixes are often already clipped, and a phone's "volume leveler" feature adds more. He also says to ask what changed, such as a new phone or a new streaming service. And if bass disappears over Bluetooth, something is faulty, because codecs strip only the highs, not the bass.

4.7 #8 A power rating is a limit, not a requirement, and "RMS power" is a misnomer.

> The right term is continuous average power, tested with pink noise at rated watts (full power only at 20 Hz, halving per octave), so a mid or tweeter rating depends on the test's high-pass. Compressed recordings raise the average power, the real danger; "3× RMS for headroom" is nonsense, and warranties cover defects, not abuse. RARE
>
> **Sources:** OFCAP 19 @39:01 · DIYMA 2013-08-01 · DIYMA 2021-09-20

In plain English: A speaker's power rating tells you how much it can take, like a speed limit, not how much it needs. "RMS power" is the term most people use. He says it is a misnomer and the right term is "continuous average power". The rating test plays pink noise at the rated watts, so full power occurs only at 20 Hz and halves every octave up. A midrange or tweeter is tested through a high-pass filter (one that blocks lower notes), so its rating depends on where that filter was set. He says the real danger is heavily compressed (always-loud) recordings, which raise the average power. Many people believe you should buy an amp three times the speaker's RMS rating "for headroom"; he calls that nonsense. He adds that warranties cover defects, not abuse.

4.7 #9 The volume control is the protection, and a customer's bass boost can undo a correct install.

> A competent user trades loudness against peak distortion ("carefully monitored", in pro-audio terms) instead of setting levels on −10 dB tones. His case: a 1970s-disco fan with no sub-level control boosted bass ~12 dB at the head unit, clipped, burned tweeters and wrecked midbass; the fix was a higher, pro-sound-style crossover. RARE
>
> **Sources:** DIYMA 2021-07-10 · CMA Connected Oct 2021 @43:34

In plain English: His point is that the person at the volume knob is the real protection. A competent listener turns up until the loud peaks start to distort, then backs off. Pro-audio people call that "carefully monitored" use. The alternative he rejects is setting levels on test tones recorded 10 dB below maximum. His example: a dealer installed a system "correctly", yet the customer kept blowing speakers. The customer played 1970s disco loud. Those recordings have little bass, and the car had no separate sub-level control. So he boosted bass about 12 dB at the head unit. That clipped the signal, burned the tweeters and wrecked the midbass drivers. The fix was to raise the crossover point and set the system up like a pro sound system. That meant accepting that the bass might no longer seem to come from the front, which this customer didn't care about.

4.7 #10 Amp power isn't what makes a car "dynamic".

> His 2010 Volvo S60 ran everything but the subs on 20 W chip amps; a 2013 24-channel car used ~30 W × 23 plus 500 W on subs; judges called his 18 W car "so dynamic". RARE
>
> **Sources:** DIYMA 2010-02-16 · DIYMA 2013-04-27

In plain English: "Dynamic" describes a system whose loud moments jump out with punch. Many people assume that takes big amplifiers. His own cars say otherwise. His 2010 Volvo S60 ran every speaker except the subs on small 20 W "chip" amps. A 2013 car with 24 channels used about 30 W on each of 23 channels, plus 500 W on the subs. And judges described his 18 W car as "so dynamic".

4.7 #11 Efficiency vs power.

> 2021: efficiency matters more (at ~88 dB/1 W/1 m only ~1% becomes sound; 16,000 W systems need heavy coils that cut efficiency further). **Conflict:** 2013 "Efficiency is overrated. Power is cheap." — said about using separate amp channels rather than resistors to level-match drivers.
>
> **Sources:** OFCAP 19 @33:07 · DIYMA 2013-11-28

In plain English: Efficiency (sensitivity) is how much sound a speaker makes from each watt. "88 dB/1 W/1 m" means it plays 88 dB from one watt, measured one metre away. In 2021 he said efficiency matters more than power. At that sensitivity only about 1% of the power becomes sound and the rest becomes heat. Huge 16,000 W systems also need heavy voice coils, which lower efficiency further. The tag flags an apparent conflict. In 2013 he wrote "Efficiency is overrated. Power is cheap." That line was about something narrower: giving each speaker its own amp channel to balance levels, instead of wasting power through resistors. Both statements are reported here as he made them.

4.8 Gain structure: DSP input, digital levels, amplifier gains

What this part is about: "Gain structure" means setting the level controls along the signal chain (radio, then DSP, then amplifiers) so the system plays loud without clipping and without audible hiss. This part covers the DSP's input setting, digital levels and amp gain knobs, his full step-by-step method, and how his advice changed over the years.

4.8 #1 The DSP input-sensitivity control has "ONE function": keep the A/D converter from clipping at maximum volume.

> Set it at or above the source's real maximum (a 4 V radio → 4–5 V, never lower); a 4 V radio into a 1 V setting allows ~12 dB of clipping nothing downstream can remove. Set it for full volume, not your usual level, above all in customers' cars. RARE
>
> **Sources:** Audiofrog blog Mar 2021 · DIYMA 2021-08-04 · DIYMA 2021-08-03

In plain English: A DSP's input-sensitivity setting tells it how strong a voltage to expect from the radio, so it can fit that signal into its A/D converter (the part that turns the voltage into numbers). He says this control has only one job: stop that converter from clipping when the radio is at full volume. So set it at or above the radio's real maximum output. If the radio makes 4 volts, set 4–5 V, never lower. If you set 1 V instead, the top of the signal (about 12 dB of it, since 4 V is four times 1 V) gets clipped. He explains that this distortion gets digitized permanently, so nothing later in the chain can remove it. Set it for full volume, not the level you usually listen at, above all in a customer's car.

4.8 #2 Find the source's true maximum by measuring its response first.

> Aftermarket radio with tone controls flat: a 1 kHz tone is fine. Factory radio or amp: measure the response (sound card and REW), find the loudest frequency and read the voltage there with a 0 dB sine and a true-RMS meter; a USB interface with balanced inputs plus an attenuator works.
>
> **Sources:** DIYMA 2021-08-04 · DIYMA 2021-08-05 · DIYMA 2013-09-17

In plain English: To set the DSP input correctly (point #1), you need to know the most voltage your source can put out. With an aftermarket radio and its bass and treble controls set flat, a 1 kHz test tone gives a fair reading. A factory radio or factory amp is different: it applies its own equalization, so it may be loudest at some other frequency. There, first measure its frequency response (how loud it plays at each frequency) using a computer sound card and REW (Room EQ Wizard, a free measurement program). Find the loudest frequency. Then play a full-level (0 dB) test tone at that frequency and read the voltage with a true-RMS multimeter, a type that reads AC voltage accurately. He says a USB audio interface with balanced inputs, plus an attenuator (a part that safely lowers the voltage), also works as the measuring tool.

4.8 #3 In a digital system err toward never clipping, and leave headroom for bass EQ.

> A little analog clipping is inaudible; a little digital clipping "sounds terrible". Two inputs mixed at 50% sum to 0 dBFS, and with every level at 100% the bass boosts clip the DSP; start DSP levels around −6 to −12 dB. 16-bit/44.1–48 kHz is enough if levels are right. RARE
>
> **Sources:** DIYMA 2021-08-04 · Audiofrog blog Dec 2020

In plain English: Inside a DSP the signal is numbers, and there is a hard ceiling called 0 dBFS ("full scale"). He says a little clipping in an analog circuit is inaudible, but a little digital clipping "sounds terrible". So in digital gear, lean toward never clipping. Levels add up quickly: two inputs each mixed in at 50% already add up to the full-scale maximum. If every level is at 100% and you then boost bass with the equalizer (EQ), the DSP clips. His advice is to start DSP levels around 6 to 12 dB below maximum. He adds that 16-bit audio at 44.1–48 kHz (CD-type quality) is enough, as long as levels are set right.

4.8 #4 His step-by-step gain procedure (2022 version), and how it evolved.

> The aim is to balance hiss, output and clipping, without scopes and meters on every channel. RARE
>
> 1. Set the DSP's analog input sensitivity at or above the source's maximum (4 V radio → 4 V or higher).
> 2. Sub only: 0 dB sine, DSP sub level and input at 0 dB, raise the sub amp gain until it just clips; then put the sine and scope away. Tweeters never get a sine (disconnect them for any sine-wave step).
> 3. Turn every DSP input and output level down 6 dB.
> 4. Crossovers and delays set by the numbers; with pink noise, bring the other amps' gains toward the target curve using level only, tweeter amp lowest.
> 5. EQ to the target and listen.
> 6. Raise all DSP outputs equally until one reaches 0 dB; check for clipping or noise.
> 7. Raise the DSP inputs toward 0 dB.
> 8. Raise every amp gain by the same amount while watching the RTA; at the first noise or distortion you've found the maximum.
>
> History: 2008 "go buy a scope" (Vmax = √(P × R), e.g. ~26.5 V for 175 W into 4 Ω, on a −10 dB sine). 2009–2015: deliberate overlap (point 5). 2014: no scope in 15 years; by ear. 2016: radio at ¾, raise the tweeter amp until distortion, back off, then lower until hiss vanishes at zero volume. 2018: amp gains per the DSP manual, tune, raise all gains until a little tweeter hiss, back off. Title-attributed DIYMA guide: DSP levels start at −12 dB; sub ~+6 dB if one input feeds it. 2021: with factory sources use the DSP's high-level input and skip the "sine at ¾ volume" routine. **Changed view:** scope and −10 dB tones (2008–2015) → sine on the sub only, the rest by pink noise and RTA (2018–2022).
>
> **Sources:** DIYMA 2022-03-03 · DIYMA 2008-10-27 · DIYMA 2018-05-04

In plain English: This is his 2022 recipe for setting levels in a system with a DSP and separate amplifiers. The goal is a balance between hiss, loudness and clipping, without putting an oscilloscope (a screen that draws the signal's wave shape) and meters on every channel.

Step 1. Set the DSP's input sensitivity at or above the most the radio can put out, so the DSP's converter can never clip (see #1 and #2).

Step 2. This step is for the subwoofer only. Play a full-level (0 dB) sine test tone with the DSP's sub level and input level at maximum (0 dB). Turn up the sub amp's gain until it just starts to clip, then stop. After that, put the test tones and scope away. Tweeters must never be fed a sine tone, so disconnect them during any step that uses one; he says setting tweeter and midrange levels with sine tones puts the speakers at unnecessary risk.

Step 3. Turn every DSP input level and output level down by 6 dB. In line with #3, this leaves headroom inside the DSP for EQ boosts.

Step 4. Set the crossovers (which frequencies go to which speaker) and the delays "by the numbers", meaning to planned values (for the delays, measured distances), not by ear. Then play pink noise and set the other amps' gains so the measured balance gets close to the target curve (the tonal balance you're aiming for), using level alone, with the tweeter amp's gain set lowest.

Step 5. Use EQ to match the target curve, then listen.

Step 6. Raise all the DSP output levels by the same amount until one of them reaches 0 dB (its maximum). Check for clipping or noise. He says to stop here if it's already loud enough.

Step 7. If not, raise the DSP input levels toward 0 dB, backing off if noise or clipping appears.

Step 8. If it is still not loud enough, turn every amp's gain up by the same amount while watching an RTA (real-time analyzer, a live display of level at each frequency), so the tonal balance stays the same. When you first hear noise or distortion, you've found the system's maximum. He says going louder than that means accepting some noise or distortion.

The "History" lines show how his method changed. In 2008 he told people to buy a scope and set gains on a −10 dB sine, working out the amp's full-power voltage with a formula (about 26.5 V for 175 W into 4 Ω). From 2009 to 2015 he advised deliberate overlap (see #5). In 2014 he said he hadn't used a scope in 15 years and set gains by ear. In 2016: radio at three-quarter volume, raise the tweeter amp until it distorts, back off, then lower it until hiss disappears with the volume at zero. In 2018: set amp gains per the DSP's manual, tune, then raise all gains until a little tweeter hiss appears and back off. A DIYMA forum guide whose thread title credits him (the research couldn't confirm who actually posted it) starts DSP levels at −12 dB, with the sub about 6 dB higher if one input feeds it. In 2021 he said that with factory sources you should use the DSP's high-level (speaker-level) input and skip the "sine at ¾ volume" routine. The [Changed view] tag sums it up: scope and −10 dB tones (2008–2015) gave way to a sine on the sub only, with everything else set by pink noise and an RTA (2018–2022).

4.8 #5 Gain overlap (deliberate clipping headroom) was once his advice; he dropped it for DSP systems.

> 2009–2015: set amps to clip on a −10 dB tone ("YOU WANT SOME CLIPPING!"), ~9–10 dB of overlap (2 V head unit → ~600 mV), since transient clipping is hard to hear; JBL's GTO "EZ" −10 dB disc built this in. Constant: never clip the A/D or head unit. **Changed view:** 2021, overlap "was never appropriate" at the DSP input, and −10 dB tones on HF amps license tweeter-killing clipping.
>
> **Sources:** DIYMA 2009-08-13 · DIYMA 2015-10-15 · Audiofrog blog Mar 2021

In plain English: "Overlap" means setting the amp so it reaches full power, and clips, before the signal feeding it reaches its own maximum. From 2009 to 2015 he recommended it on purpose: set the amps to clip on a test tone recorded 10 dB below maximum, which gives about 9–10 dB of overlap. For example, with a radio that puts out 2 V, set the amp's sensitivity to about 600 mV (0.6 volts). His reasoning then was that brief clipping on short peaks is hard to hear, and quieter recordings still play loud. JBL's GTO "EZ" amps came with a −10 dB test disc that built this in. One rule never changed: never clip the A/D converter or the head unit. In 2021 he changed his view. He said overlap "was never appropriate" at the DSP input. He also said −10 dB tones on treble amps allow clipping that kills tweeters.

4.8 #6 Starting point: amp gains around 2 V.

> With MS-8 or any DSP he repeatedly says ~2 V (JBL MS amps: "Low" switch, pot fully counter-clockwise); an 8 V setting guarantees the amp never clips and wastes ~12 dB.
>
> **Sources:** DIYMA 2015-04-24 · DIYMA 2013-05-25

In plain English: An amp's gain knob is usually marked in volts: the input voltage that drives the amp to full power. With JBL's MS-8 (a specific JBL car processor) or any other DSP, he repeatedly suggests starting around 2 V. On JBL MS-series amps, that means the "Low" switch position with the knob turned fully counter-clockwise. Setting 8 V means the amp needs 8 volts in to reach full power. Fed a signal of around 2 V, it never gets there, so it never clips, but you throw away about 12 dB of its output (8 V is four times 2 V).

4.8 #7 The amp gain knob is a level-matching control, and low gain isn't automatically low noise.

> If noise arises after the pot, low gain worsens the signal-to-noise ratio; test with RCAs unplugged and gain at max (hiss means the amp's own preamp). A tweeter that plays 3–6 dB louder than the woofer (Audiofrog's do) lets that channel's gain, and its hiss, come down. RARE
>
> **Sources:** DIYMA 2010-09-11 · DIYMA 2016-06-06 · DIYMA 2017-10-01

In plain English: Many people believe turning an amp's gain down always means less noise. He says the gain knob is really for matching levels between channels, and a low setting isn't automatically quieter. If an amp's hiss is created after the gain control (the "pot"), turning the gain down lowers the music but not the hiss, so the signal-to-noise ratio gets worse. His test: unplug the RCA cables (the signal leads) and turn the gain all the way up. If you hear hiss, it comes from the amp's own preamp stage, and a lower setting will help. He adds that a tweeter that plays 3–6 dB louder than the woofer, as Audiofrog's tweeters do, lets you turn that channel's gain down, and its hiss with it.

4.8 #8 Line drivers are "possibly the most unnecessary component in all of car audio", and fat RCAs are useless too.

> They made sense with ~100 mV head units and single-ended amps; into a differential input, an 8 V line driver is just 6 dB of overlap. When one "fixed" noise, its ground break did it; amp input impedance makes heavy-gauge RCAs pointless. RARE
>
> **Sources:** DIYMA 2010-06-01 · DIYMA 2011-12-27 · DIYMA 2021-07-10

In plain English: A line driver is a small box that boosts the radio's signal voltage before it reaches the amps. He calls it "possibly the most unnecessary component in all of car audio". It made sense when radios put out only about 100 mV (a tenth of a volt) and amps had "single-ended" inputs, the older kind that use the cable's outer shield as the ground return. A "differential" input instead measures the difference between two wires and ignores noise common to both. Into a differential input, he says, an 8 V line driver just adds 6 dB of overlap (signal beyond what the amp needs; see #5). When a line driver seemed to cure noise, he says the cure came from its ground isolation, not its extra voltage. He also says thick, heavy-gauge RCA cables are pointless: amp inputs have a high input impedance (resistance to current), so they draw almost no current and wire thickness doesn't matter.

4.8 #9 DSP amplifiers have no analog gain after the DAC, so EQ cuts cost output you can't win back.

> The input is attenuated to fit the A/D (~1–4 V), costing SNR; the D/A can't exceed 0 dBFS, so after tuning cuts a "1,000 W" DSP amp may deliver ~10 W. For high-end systems use a separate DSP and amps with analog input-sensitivity controls; with a DSP amp, EQ dips up as well as cutting peaks. RARE
>
> **Sources:** Absolute Electronix 2023 @33:35 · DIYMA 2022-03-03

In plain English: A "DSP amplifier" is one box containing both a processor and an amplifier. The signal goes in, is shrunk to fit the A/D converter, whose ceiling is about 1–4 V, and that shrinking costs some signal-to-noise ratio (SNR, how far the music sits above the hiss). Then a DAC, or D/A (digital-to-analog converter), turns the numbers back into a voltage, and it can never go above 0 dBFS. In these boxes there is no adjustable analog gain after that point. So every EQ cut you make during tuning lowers the maximum output for good. In his example, after tuning cuts a "1,000 W" DSP amp may deliver about 10 W. For high-end systems he recommends a separate DSP plus amps that have analog input-sensitivity controls. If you do use a DSP amp, he says to EQ by boosting dips as well as cutting peaks.

4.8 #10 "All-digital" signal chains waste amplifier power.

> Nothing exceeds 0 dBFS and cars need 6–12 dB more bass, so everything else must be attenuated; multi-thousand-watt digital systems end up sounding like 300 W ones. An analog chain with balanced outputs and differential inputs is "just fine". RARE
>
> **Sources:** DIYMA 2017-07-05

In plain English: In an "all-digital" chain the signal stays as numbers until the very end, with no analog level controls along the way. Nothing can go above the digital ceiling (0 dBFS). Cars need 6–12 dB more bass than the rest of the range, so everything else has to be turned down to make room for that bass boost. The result, he says, is that systems with thousands of watts end up sounding like 300 W systems. Many people assume all-digital is automatically better. He says an analog chain using balanced outputs and differential inputs (wiring that cancels noise picked up along the cable) is "just fine".

4.8 #11 Know your volts and decibels.

> Scope readings are peak (4 V peak ≈ 2.83 V RMS); ~4.24 V RMS is the most a 12 V head unit makes without a switching supply. Use 20 log for voltage and 10 log for power (a 6 dB cut is 75% less power); splitting a preamp signal doesn't lower its level; non-true-RMS meters keep rising into clipping.
>
> **Sources:** DIYMA 2013-07-21 · DIYMA 2013-09-17 · DIYMA 2017-10-11

In plain English: These are the basic numbers he wants installers to get right. An oscilloscope shows a wave's peak voltage. A multimeter shows "RMS" voltage, a kind of effective average; for a sine wave, RMS is about 0.707 times the peak, so 4 V peak is about 2.83 V RMS. He says about 4.24 V RMS is the most a head unit running on 12 volts can make without a switching power supply (a circuit that raises the voltage). To turn voltage ratios into decibels you use "20 log"; for power ratios you use "10 log". So a 6 dB cut is half the voltage but 75% less power. Splitting one preamp

signal to feed several amps doesn't lower its level. And a meter that isn't "true-RMS" keeps showing rising numbers even after the signal starts clipping; he says this makes setting gains with an ordinary multimeter imprecise.

4.9 Amplifier specs and design: what matters, what doesn't

What this part is about: Which numbers on an amplifier's spec sheet actually matter, which are marketing, and what really makes one amp sound different from another. It ends with details of his own company's amplifiers.

4.9 #1 Damping factor is a marketing spec; above ~10 it's meaningless.

> The voice coil's own resistance (~3.2–4 Ω) sits in series with the amp's output impedance and sets the current, so DF 1000 moves Qts from 0.707 to 0.708. Output impedance matters only when it nears the coil's DCR, where it tilts response; that is tube "warmth". RARE
>
> **Sources:** Audiofrog blog Dec 2020 · DIYMA 2017-07-12 · DIYMA 2021-05-08

In plain English: "Damping factor" (DF) is a spec that claims to show how tightly an amp controls a speaker's cone. Many people believe a huge number like 1000 means tighter bass. He calls it a marketing spec that is meaningless above about 10. Resistance is measured in ohms (Ω). The voice coil's own wire resistance, its DCR (about 3.2–4 Ω), sits in line with the amp's tiny internal resistance (its "output impedance"), and it is the coil's resistance that sets the current. So going to DF 1000 changes the speaker's Qts (a number describing how damped, or controlled, its resonance is) only from 0.707 to 0.708, a negligible change. The amp's output impedance matters only when it gets close to the coil's DCR. Then it tilts the frequency response, and he says that is the "warmth" people hear from tube amps.

4.9 #2 Slew rate is already covered by the power-bandwidth (THD) spec; compare distortion at ~1 W.

> A slew limit shows up as HF distortion in the THD figure. Tweeters live around a watt, and some amps distort more at very low power than at rated power. RARE
>
> **Sources:** DIYMA 2017-07-07 · DIYMA 2021-05-18

In plain English: "Slew rate" is how fast an amp's output voltage can change. Some buyers treat it as a separate quality measure. He says it is already covered by the THD (total harmonic distortion) spec measured across the full frequency range at rated power. If an amp can't change its voltage fast enough, that shows up as extra distortion at high frequencies in the THD number. The more useful thing to compare is distortion at about 1 watt. That's because tweeters live around a watt, and some amps distort more at very low power than at full rated power.

4.9 #3 "Class D is only for bass" is backwards; class D's real car problem is FM interference.

> In Harman's 1996 CES blind demo (Infinity Beta class D vs a Krell) only 3–4 trained listeners heard a difference, in long bass notes. Class AB and D differ only when audibly clipped, so judge implementations ("all marinara is terrible because you tried Ragu"). RARE Conflict: ~800 demos/3 listeners (2022) vs ~1,500 demos/4 listeners (2023).
>
> **Sources:** CMA 2023 @25:04 · CMA 2022 @29:01 · DIYMA 2022-03-26

In plain English: "Class AB" and "class D" are two ways of building an amplifier. Class D switches its output transistors on and off very fast, which makes it small and efficient. Many people believe class D is only good enough for subwoofers. He says that's backwards, and that class D's real problem in cars is interference with FM radio reception. His evidence is a 1996 blind demo by Harman (the company behind JBL and Infinity) at the CES trade show. It compared an Infinity Beta class-D amp with a Krell, a high-end amp. Only 3–4 trained listeners heard a difference, and only on long bass notes. He says class AB and class D sound different only when audibly clipping, so judge each amp on its design, not its class. Hence his line "all marinara is terrible because you tried Ragu". The [Conflict] tag notes that he gave different counts in two interviews: about 800 demos and 3 listeners (2022), versus about 1,500 demos and 4 listeners (2023).

4.9 #4 Class-D treble depends on the load.

> The passive output filter is designed for one impedance: with a ~20 kHz filter, 2 Ω loads roll off highs and 8 Ω or rising tweeter impedance peak. Audiofrog puts it at ~40 kHz, matched to its low-inductance tweeters (flat almost to 30 kHz); big HF differences between amps usually come from output filters. RARE
>
> **Sources:** Absolute Electronix 2023 @29:58 · CMA 2022 @35:07 · DIYMA 2021-05-18

In plain English: A class-D amp has a small filter on its output to remove the switching noise. That filter is designed to behave correctly with one particular load impedance (the speaker's resistance to the signal). If the filter is set around 20 kHz, a 2 Ω load makes the highest treble roll off (drop), while an 8 Ω load, or a tweeter whose impedance rises at high frequencies, makes the treble peak (rise). He says Audiofrog places this filter at about 40 kHz and matches it to its own low-inductance tweeters, which stay flat almost to 30 kHz. His point is that when two amps sound very different in the treble, the output filters are usually the reason.

4.9 #5 You can't audition an amp by itself.

> What people hear is output impedance interacting with the speaker, or clipping; "one unclipped watt is one unclipped watt". Buy expensive amps for features, longevity or clipping behavior; GaN brings smaller size and lower HF distortion, but silicon is fine. RARE
>
> **Sources:** CMA 2023 @19:05 · DIYMA 2013-05-19 · DIYMA 2022-03-26

In plain English: When people compare amps by ear, he says they are really hearing the whole system: the amp's output impedance interacting with that particular speaker, or the amp clipping. Apart from that, "one unclipped watt is one unclipped watt". So the reasons to pay more for an amp are its features, how long it lasts, or how it behaves when it clips. GaN (gallium nitride, a newer type of output transistor) brings smaller size and lower distortion at high frequencies, but he says silicon is fine.

4.9 #6 Low noise and low-level distortion matter more than headline watts, and noise is board layout.

> Tiny class-D amps hiss or whine because the supply transformer's field sits near preamp parts; big amps "solve" it with distance. Audiofrog uses a 4-layer board with a ground-shield layer, audio parts where the transformer's field is weakest, and a chassis that shields FM. RARE
>
> **Sources:** CMA 2023 @22:05 · Absolute Electronix 2023 @26:01 · CMA 2022 @44:00

In plain English: He says a quiet amp with low distortion at low volume matters more than a big wattage number. Car amps contain a power-supply transformer that raises the car's 12 volts, and it gives off a magnetic field. In tiny class-D amps that field sits right next to the delicate preamp (small-signal) parts, which causes hiss or whine. Big amps "solve" this simply by putting distance between them. His company, Audiofrog, instead uses a circuit board with four layers, one of them a ground layer that acts as a shield. It places the audio parts where the transformer's field is weakest, and uses a metal case that shields against FM radio interference. So in his view, noise comes down to how the circuit board is laid out.

4.9 #7 Channel separation beyond ~60 dB is irrelevant, and separate left/right amps buy nothing.

> 60 dB is a millionth of the power, while acoustic crosstalk in a car is only 6–12 dB; for separation "a car wash is more likely" to help. A full-range bridged amp causes no image-damaging phase shift.
> RARE
> **Sources:** DIYMA 2022-03-26 · DIYMA 2021-11-05 · DIYMA 2021-09-15

In plain English: "Channel separation" measures how little of the left channel leaks into the right inside the amp. Some people buy separate amps for left and right to improve it. He says anything beyond about 60 dB is irrelevant: 60 dB of separation means the leak is a millionth of the power. Meanwhile, in a car the sound itself leaks. Sound from one side of the car reaches the other side, directly and by reflecting off the windows, only 6–12 dB quieter. That acoustic "crosstalk" dwarfs anything the amp does, so he jokes "a car wash is more likely" to help. He also says a full-range bridged amp (two channels combined to drive one speaker) causes no phase shift that would damage the stereo image (the sense of where instruments are placed).

4.9 #8 Car amps are voltage sources: bridging quadruples power, and halving the load does not double it.

> One channel at 10 V peak into 4 Ω is 25 W; bridged, the speaker sees 20 V, 100 W. Amps "know" only current; past their limit modern amps sag the supply rails rather than shut down, so "set gains under load" matters only deep into clipping.
> **Sources:** DIYMA 2012-05-12 · DIYMA 2022-02-10 · DIYMA 2021-11-14

In plain English: A "voltage source" holds its output voltage steady and lets the current change with the load. Power depends on voltage squared, so doubling the voltage gives four times the power. One channel swinging 10 V peak into a 4 Ω speaker makes 25 W. Bridged, the two channels push in opposite directions, so the speaker sees 20 V and gets 100 W. Many people believe that halving the speaker's impedance (say 4 Ω to 2 Ω) doubles an amp's power. He says it does not, and puts this down to real-world inefficiencies. He also says amps "know" only current. Pushed past their limit, modern amps let their internal supply voltage (the "rails") sag rather than shut down. So the common advice to "set gains under load" matters only when the amp is driven deep into clipping.

4.9 #9 Sanity-check power claims with the fuse and the heatsink.

> A 20 A fuse and a 2,000 W rating don't go together; some amps' thermal rollback kicks in within minutes, so "150 W" acts like 75 W. Designers now save aluminum by rolling back supply voltage

> when hot. RARE
>
> **Sources:** DIYMA 2011-12-16 · CMA 2022 @1:09:04 · DIYMA 2022-01-14

In plain English: You can check whether an amp's power claim is believable. An amp running on about 12 volts can draw only as much current as its fuse allows, and a 20 A (amp) fuse simply can't support a 2,000 W rating. The heatsink (the metal body that sheds heat) is another clue. Some amps have "thermal rollback": when they get hot, they cut their own power, sometimes within minutes. So an amp rated "150 W" may behave like a 75 W amp. He says designers now save on aluminum (smaller heatsinks) by lowering the supply voltage when the amp gets hot.

4.9 #10 Built-in amp crossovers are "broccoli on the buffet": defeat them under a DSP.

> Set the A150.4D filter to "None" and open the A600.1D fully (HP 5 Hz, LP 500 Hz). Its filters are 12 dB/oct because 24 dB pot filters need 8-gang pots that never track; with 12 dB amp filters, alternate polarity between bands and keep an octave between high- and low-pass.
>
> **Sources:** DIYMA 2013-10-31 · A-Series manual · DIYMA 2022-02-28

In plain English: Many amps have crossover filters built in. He calls them "broccoli on the buffet": cheap to include, and you can take them or leave them. When a DSP is doing the crossovers, switch the amp's filters off. On Audiofrog's A150.4D, set the filter to "None". On the A600.1D, open it fully: high-pass (HP) at 5 Hz and low-pass (LP) at 500 Hz, its widest settings, so the DSP does the filtering. He explains that these amps' filters are 12 dB per octave (a gentle slope). A steeper 24 dB filter adjusted by knobs would need "8-gang" pots, eight controls on one shaft, which never stay matched to each other. If you do use 12 dB amp filters, he says to reverse the speaker polarity (wiring +/−) on alternate bands, and to keep an octave between the high-pass and low-pass points.

4.9 #11 Audiofrog A-series (product-specific): two small models, no DSP inside, build any system.

> A150.4D (~8 × 4 × 1.25 in): 4×100 W at 4 Ω, 4×150 W at 2 Ω, 2×300 W bridged (dyno ~175 W × 4 at 2 Ω); A600.1D: 450 W at 4 Ω, 600 W at 2 Ω (dyno ~750 W); 60 A fuse each. An active 3-way front is one A150.4D on tweeters and mids plus one bridged for midbass; RLC-Sync makes all sub amps follow the lowest-set one.
>
> **Sources:** A150.4D page · A600.1D page · Absolute Electronix 2023 @23:06

In plain English: This point is about specific products from Audiofrog, Andy's company. The A-series has just two small amps with no DSP inside, meant to be combined into any system. The A150.4D is about 8 × 4 × 1.25 inches (roughly 20 × 10 × 3 cm). It is a 4-channel amp rated 100 W per channel at 4 Ω, 150 W per channel at 2 Ω, or 300 W × 2 bridged; on a dyno (a test bench) it made about 175 W × 4 at 2 Ω. The A600.1D is a 1-channel amp rated 450 W at 4 Ω and 600 W at 2 Ω, and measured about 750 W. Each uses a 60 A fuse. An "active 3-way" front (tweeter, midrange and midbass each on their own amp channel) takes one A150.4D for the tweeters and mids plus a second, bridged, for the midbass. RLC-Sync is a feature of its remote level control: with several sub amps, all of them follow the one set lowest.

4.10 Noise: hiss, whine, ground loops and pops

What this part is about: How to tell the different kinds of unwanted noise apart (steady hiss, engine-speed whine, turn-on pops), how to find which box is making them, and which fixes actually work.

4.10 #1 Ground loops make whine; hiss is almost never a ground problem.

> Hiss that stays with the RCAs unplugged is the amp; hiss that rises with head-unit volume is the head unit; noise that moves sides when you swap both RCAs and speaker leads is the amp. Tying RCA shields together fixes whine, not hiss. RARE
>
> **Sources:** DIYMA 2015-09-16 · DIYMA 2015-09-27 · DIYMA 2022-03-01

In plain English: Hiss is a steady "shhh" sound. Whine is a tone, often one that rises and falls with engine speed. A ground loop happens when two connected pieces of equipment see slightly different "ground" (zero-volt reference) voltages, and the difference leaks into the audio. Many people blame every noise on grounding. He says ground loops make whine, but hiss is almost never a ground problem. His checks: if the hiss stays when the RCA signal cables are unplugged from the amp, the amp is making it. If the hiss gets louder as you turn up the head unit, the head unit is making it. If you swap both the left/right RCAs and the left/right speaker leads at the amp and the noise moves to the other side, the amp is the cause. Tying the RCA cables' shields (outer conductors) together can fix whine, but not hiss.

4.10 #2 Ground loops are designed into equipment, and differential inputs largely ended the ground-loop era.

> The loop is a difference between chassis and audio ground inside the boxes, so chasing ground points is often wasted effort. Threads list re-grounding, 0-gauge cable, the "Big 3" and new RCAs with the noise unchanged: "that's not the problem anymore". RARE
>
> **Sources:** DIYMA 2013-03-09 · DIYMA 2021-09-24 · DIYMA 2021-11-03

In plain English: He says a ground loop is built into the equipment itself: it comes from a voltage difference between the metal case (chassis) ground and the audio ground inside the boxes. So moving ground wires to new bolts in the car is often wasted effort. A "differential" input measures the difference between two signal wires and ignores noise that appears on both, and he says such inputs largely ended the ground-loop era. He points to forum threads where people re-grounded everything, fitted 0-gauge (very thick) power cable, did the "Big 3" (a common upgrade of the car's three main charging cables) and bought new RCA cables, and the noise didn't change. His verdict on grounding points: "that's not the problem anymore".

4.10 #3 Ground-loop triage.

> 2008: unplug the RCAs at the amp (if noise stays, a grounding block won't help); touch the amp's RCA shield to the radio case (if noise goes, make that wire permanent); otherwise fit transformer isolators. 2015: if an amp's input shields read 0 Ω to its power ground it is single-ended and built to make loops, so avoid buying it. 2021: on a Helix, try the 200 Ω input-ground setting. RARE
>
> **Sources:** DIYMA 2008-12-06 · DIYMA 2015-04-18 · DIYMA 2021-11-03

In plain English: "Triage" means sorting out the problem in order. These are his tips from three different years. 2008: first unplug the RCA signal cables at the amp. If the noise is still there, a grounding block (a terminal block for sharing ground wires) won't help. Next, touch the outer shield of the amp's RCA cable to the radio's metal case. If the noise goes away, make that connection permanent with a wire. If neither works, fit transformer isolators (small boxes that pass the audio through a transformer and break the ground connection). 2015: use an ohmmeter (a meter that measures resistance). If an amp's input shields read 0 Ω to its own power ground, it has "single-ended" inputs that are built to make loops, so he says avoid buying it. 2021: on a Helix (a brand of DSP), try its 200 Ω input-ground setting.

4.10 #4 The best cure: a floating speaker-level output into a differential input — no ground connection, no loop.

> Its higher voltage also rejects radiated noise; a high-to-low converter (LOC) that cuts voltage tenfold often creates the noise, and shops then sell a line driver to win it back. Optical's one real advantage, radiated-noise immunity, is matched "damn close". RARE
>
> **Sources:** DIYMA 2010-06-01 · DIYMA 2009-08-23 · DIYMA 2013-11-22

In plain English: A radio's speaker outputs are often "floating", meaning not tied to ground. He says the best cure for ground loops is to feed such a speaker-level output straight into an amp or DSP input that is "differential" (it reads the difference between two wires). With no ground connection, there is no loop. The speaker-level signal's higher voltage also helps it stand out against noise picked up through the air ("radiated noise"). A common alternative is a high-to-low converter, or LOC (line output converter), which cuts the voltage to RCA level. He says an LOC that cuts the voltage tenfold often creates the noise problem, and shops then sell a line driver to boost the voltage back up. Some people prefer optical (light-based) digital connections. He says their one real advantage, immunity to radiated noise, is matched "damn close" by this floating-output method.

4.10 #5 Match cable to input: twisted pair for differential inputs, coax only for single-ended.

> Ohmmeter test: RCA shield to the amp's power ground near 0 Ω means single-ended (keep any LOC); hundreds to thousands of ohms means differential (feed it directly). Either mismatch amplifies noise; which RCA rejects noise best depends on the input circuit, not the cable. RARE
>
> **Sources:** CMA 2022 @50:01 · DIYMA 2021-09-17 · DIYMA 2021-11-26

In plain English: There are two common signal-cable designs. "Twisted pair" is two wires twisted together; "coax" is one centre wire inside a surrounding shield, like a typical RCA cable. He says twisted pair suits differential inputs, and coax suits only single-ended inputs. To find out which input an amp has, use an ohmmeter between the RCA jack's outer shield and the amp's power-ground terminal. Near 0 Ω means single-ended, so keep any LOC (line output converter) you're using. Hundreds to thousands of ohms means differential, so you can feed it directly. Using the wrong cable for the input type, either way round, amplifies noise. And which RCA cable rejects noise best depends on the amp's input circuit, not on the cable brand.

4.10 #6 Transformer ground-loop isolators "aren't a band-aid"; for alternator whine try a 4700 µF cap at the alternator first.

> Put a polarized 4700 µF cap on the alternator output post, negative to chassis; if that fails, use a real transformer isolator such as the ~$20 PAC SNI-1.
>
> **Sources:** DIYMA 2013-03-19 · DIYMA 2008-08-19

In plain English: Isolators are sometimes dismissed as a band-aid, a quick patch over a bad install. He says transformer isolators "aren't a band-aid", meaning they are a legitimate fix. For alternator whine (a whine that follows engine speed, from the car's charging alternator), he suggests trying a capacitor first. A capacitor stores electrical charge and can smooth ripples in a supply. Use a polarized one (it has a marked + and − side) of 4700 µF (microfarads, its size). Connect it to the alternator's output post, with the negative side to the chassis. If that doesn't work, use a real transformer isolator, such as the PAC SNI-1, which costs about $20.

4.10 #7 Amps with built-in DSP are inherently noisier.

> Their preamp must attenuate any input to fit the A/D (~1–1.4 V RMS max) and stay very quiet doing it; know whether the gain control works before or after the A/D. If hiss falls with gain turned down and nothing connected, the amp is the source; on a tweeter-only amp, lower its gain and boost highs in the DSP. RARE
>
> **Sources:** DIYMA 2021-10-14

In plain English: An amp with a DSP built in has an A/D converter that can only accept about 1–1.4 V RMS. So whatever voltage comes in, a preamp stage first has to shrink it to fit, and it must stay very quiet while doing so. That, he says, makes these amps inherently noisier. Find out whether the amp's gain control works on the analog signal before the converter or on the digital level after it. His test: with nothing plugged in, if the hiss drops as you turn the gain down, the amp is making the hiss. If that amp only drives tweeters, turn its gain down and boost the treble in the DSP instead. He explains there is room for that boost because treble content in most music sits around −20 dB.

4.10 #8 Small DSPs need an isolating DC/DC supply; leave the MS-8 mic unplugged; whack the box for intermittents.

> Turn-on thumps and engine noise with miniDSP-class units call for a converter that isolates signal ground (JBL's MS-2 got one); the MS-8 mic is multiplexed to its inputs.
>
> **Sources:** DIYMA 2011-11-11 · DIYMA 2011-11-21

In plain English: These are three separate tips. First, small, inexpensive DSP boards (the miniDSP type) can make thumps at turn-on and pick up engine noise. He says they need a DC/DC converter, a small power supply that isolates the signal ground from the car's ground. JBL's MS-2 processor was given one for this reason. Second, on the JBL MS-8 (a specific car DSP that sets itself up using a calibration microphone), unplug the microphone when you're not calibrating. The mic is "multiplexed" (switched) into the same circuitry as the audio inputs, and he says leakage from it adds noise. Third, for noise that comes and goes, give the unit a firm whack. He says that helps reveal bad solder joints or loose connectors.

4.10 #9 Odd noise sources: leads touching metal and passive crossovers near electronics.

> A sub's tinsel lead touching its grounded basket makes a ground loop on bass notes (or sends the amp into protect). For whine in one door, check the speaker leads aren't touching metal, then move the passive crossover: at low current its coils pick up radiated noise.
>
> **Sources:** DIYMA 2011-10-16 · DIYMA 2022-01-02 · DIYMA 2021-08-26

In plain English: Two unusual causes of noise. First, a subwoofer's "tinsel leads" are the flexible braided wires that carry signal from its terminals to the moving voice coil. The metal frame (basket) is often grounded to the car through its mounting screws. If a tinsel lead touches that frame, it can create a ground loop that is heard on bass notes, or it can send the amp into "protect" mode (it shuts itself off). Second, for whine in just one door, check that the speaker wires aren't touching metal. Then move the passive crossover (the filter box of coils and capacitors) away from the car's electronics. He says that when little current is flowing, its coils pick up noise radiated through the air.

4.10 #10 Turn-on pop: find which box makes it, and know the IR2092 fix.

> Unplug the amp's inputs: if it still pops, the amp is the source (more than a few mV DC at its outputs is unacceptable; fix a no-DC pop with delayed speaker relays); if not, delay the amp's turn-on. Amps built on the IR2092 class-D driver IC have an unavoidable on/off offset: put a resistor (~10 Ω) across the tweeter protection cap, and keep that cap even in active systems in case an amp fails. RARE
>
> **Sources:** DIYMA 2011-08-20 · DIYMA 2022-03-16 · DIYMA 2021-07-06

In plain English: A turn-on pop is a thump from the speakers when the system switches on or off. First find which box makes it: unplug the amp's input cables and switch on. If it still pops, the amp is the source. In that case, measure the DC voltage at its speaker outputs: he says more than a few mV (thousandths of a volt) is unacceptable. If there's no DC and it still pops, fit relays (electrically operated switches) that connect the speakers a moment later. If it doesn't pop with the inputs unplugged, the pop comes from earlier gear, so delay the amp's turn-on instead. Amps built around the IR2092 chip (a common class-D driver part) have a small, unavoidable voltage jump at on and off. Many tweeters have a protection capacitor wired in line to block bass. His fix is to put a resistor of about 10 Ω across that capacitor. And he says to keep that capacitor even in an active system (where a DSP does the crossovers), in case an amp fails.

4.11 Wiring, fusing, grounding, batteries, caps and the Big 3

What this part is about: How to get power to an amplifier safely and well: where to connect the ground, how to test it, how to join wires, how to fuse, and whether you need thick cable, extra capacitors or electrical upgrades. His general view is that much of the common advice here is overkill.

4.11 #1 Ground the amp to body sheet metal near it, not the frame; the ground wire can be thinner than the power wire.

> The body has more cross-section than any cable and both battery and alternator bond to it; a frame on rubber bushings reduces to its bolts. Scrape paint, use a star washer; doubled 0-gauge battery

> grounds are pointless, and a shorter ground run allows a smaller gauge. RARE
>
> **Sources:** DIYMA 2008-10-17 · DIYMA 2011-08-15 · DIYMA 2021-10-04

In plain English: An amplifier needs two power connections: a positive wire from the battery, and a ground (negative) wire. In a car, the metal body itself carries current back to the battery. His point is that the body's sheet metal has more metal for current to flow through than any cable, and both the battery and the alternator (the engine-driven generator) are connected to it. A frame (the separate steel chassis under some vehicles, such as many trucks), by contrast, sits on rubber bushings, so it connects to the rest of the car only through its bolts. So ground the amp to body metal close to it. Scrape off the paint so bare metal touches, and use a star washer (a toothed washer that bites into the metal). "Gauge" is wire thickness; smaller numbers mean thicker wire, and 0-gauge is very thick. He says doubling up 0-gauge ground cables at the battery is pointless, and because the ground run is usually much shorter than the power run, it can use thinner wire. Many people believe the ground wire must always be at least as thick as the power wire.

4.11 #2 Check grounds by voltage drop, not resistance.

> A DMM can't read resistance in a circuit with current flowing; with the battery connected, measure the voltage drop across the joint.
>
> **Sources:** DIYMA 2021-07-27

In plain English: A DMM (digital multimeter) measures resistance by pushing its own small current through the part it is testing. He explains that it can't do that while the car's own current is already flowing through the circuit. So instead, leave the battery connected, set the meter to volts, and measure the voltage across the joint (one probe on each side of the connection). Any voltage lost across a joint is the sign of resistance there.

4.11 #3 Crimp, don't solder or tin, wire-to-terminal connections, including the Big 3.

> Solder near the battery invites corrosion; screw terminals should clamp bare strands; solder wicks up and stiffens speaker tinsel leads, and a hot iron can melt the plastic terminal boards in Audiofrog GS baskets. Crimp skeptics, he says, "crimp with pliers". RARE
>
> **Sources:** DIYMA 2021-07-21 · DIYMA 2021-11-26 · DIYMA 2018-06-24

In plain English: Crimping means squeezing a metal terminal tightly onto a wire with a proper tool. Soldering joins metal with melted alloy, and "tinning" means coating bare wire strands with solder. He prefers crimping, for several reasons. Solder near the battery invites corrosion. A screw terminal should clamp bare strands, so the screw compresses them and increases the contact area. On speakers, solder can creep up the thin, flexible "tinsel" leads (the braided wires that carry the signal to the moving coil) and stiffen them. And on Audiofrog GS speakers, whose baskets (frames) have plastic terminal boards, a hot iron can melt the plastic. The "Big 3" is a common upgrade of the three main charging cables. His jab at people who distrust crimps is that they "crimp with pliers", meaning with the wrong tool. Many people believe solder always makes the best connection.

4.11 #4 A second battery: ground it to the body, wire it in parallel, and skip the solenoid isolator.

> In a truck that killed six MS-8s, the solenoid dropped and reconnected the battery while cranking, and its un-diode'd coil spiked the remote line. Chain turn-on as radio REM → DSP REM-in, DSP REM-out → amps (under 1 A), so amps start after the DSP. RARE
>
> **Sources:** DIYMA 2012-01-13 · DIYMA 2012-01-14

In plain English: Some people add a second battery for a big system. His advice: ground it to the body, connect it in parallel with the first battery (positive to positive, negative to negative), and don't use a solenoid isolator, which is a heavy electrical switch that connects and disconnects the second battery. His reason is a real case: a truck killed six MS-8 processors (JBL's MS-8 was a car-audio DSP, a digital sound processor). While the engine was cranking, the solenoid dropped the battery and reconnected it. Its coil had no diode, the small part that normally absorbs the voltage spike a switching coil makes, so the spike went into the "remote" wire, the thin wire that tells audio equipment to switch on. He also gives a turn-on order: radio's remote output to the DSP's remote input, then the DSP's remote output to the amps (keeping the current drawn from that DSP output under 1 amp). That way the amps switch on after the DSP.

4.11 #5 The amp fuse protects the car, not the amp, and every extra fuse or block is a resistor.

> Fit it within 18" (45 cm) of the battery (60 A for the A-series). Customers' added fuses ahead of JBL's A6000GTi dropped enough voltage on ~800 A bass peaks to trip its protection; if the whole car dims and dies, look at the alternator–battery circuit. RARE
>
> **Sources:** DIYMA 2013-03-26 · DIYMA 2012-09-12 · A-Series manual

In plain English: A fuse is a deliberate weak link that melts if too much current flows. His point is that the main amp fuse is there to protect the car, not the amp. In the forum post behind this point, he explains that the amp's own output parts fail individually at much lower current than the fuse allows, so the fuse can't save them; what it does is stop one failure from cascading and starting a fire. Fit it within 18 inches (45 cm) of the battery; Audiofrog's A-series amps call for a 60 A fuse. He also warns that every extra fuse or distribution block acts like a resistor and loses some voltage. His example: customers added extra fuses in front of JBL's large A6000GTi amp. On bass peaks of about 800 amps, those fuses lost enough voltage to trip the amp's protection, so it shut down. And if the whole car's lights dim and the car dies, he says the problem is in the alternator–battery circuit, not in the amp.

4.11 #6 Wire-gauge and charging "requirements" assume sine waves at fuse rating; music draws far less.

> Music averages ~1/32 of peak; IASCA judges once docked him for wire that "couldn't work" while hearing it work, and a GB active system in a 2021 Sierra didn't need 0-gauge. Audiofrog lists current draw (0.9 A idle, 60 A max) but no gauge. RARE
>
> **Sources:** DIYMA 2012-10-21 · DIYMA 2021-12-22

In plain English: Charts tell you how thick a power cable and how big a charging system you "need" for a given amp. He says those charts assume the amp plays a steady test tone (a sine wave) at the full current its fuse allows. Real music is mostly quiet with short peaks; its average power is only about 1/32 of its peak. So real systems draw far less. Two examples: at an IASCA contest (a sound-quality competition), judges took points off because they said his power wire "couldn't

work", while they were listening to it work. And a GB (Audiofrog) active system in a 2021 GMC Sierra truck didn't need 0-gauge (very thick) cable. Audiofrog lists its amps' current draw (0.9 A at idle, 60 A maximum) but not a wire gauge. Many people believe you must size wire straight from the amp's fuse rating.

4.11 #7 Stiffening caps and electrical upgrades are usually unnecessary.

> A 2-farad cap isn't needed at 1,000 W (one kept a member's battery from charging), and he told another member to skip an electrical upgrade. Alternator and battery rules aren't absolute: a Harman trainer's "two 2,500 W amps will shut the car down" demo met an attendee running two on a 40 A alternator. He gives no further view on the Big 3 beyond "crimp it"; silver wire isn't worth trying. RARE
>
> **Sources:** DIYMA 2021-08-02 · DIYMA 2011-11-25 · DIYMA 2021-01-13

In plain English: A "stiffening cap" is a large capacitor fitted near the amp to supply quick bursts of current. He says these, and electrical upgrades in general, are usually not needed. A 2-farad cap isn't needed for a 1,000-watt system, and one actually stopped a forum member's battery from charging. He told another member to skip an electrical upgrade. He adds that alternator and battery rules aren't absolute: a Harman trainer told a class that two 2,500-watt amps would shut a car down, and one attendee was running two of them on a 40-amp alternator. On the Big 3 upgrade, the only view he gives is "crimp it". Silver wire, he says, isn't worth trying. Many people believe every big system needs a capacitor and a charging upgrade.

4.11 #8 Terminations and passive-network safety.

> Crimp ferrules with the proper tool (the terminal screw is not a crimper). Never drive a 2nd-order or steeper passive network without its tweeter connected: with the tweeter open, the shunt coil becomes a near-dead short at the network's resonance.
>
> **Sources:** Audiofrog blog Dec 2020 · DIYMA 2018-01-13

In plain English: A ferrule is a small metal sleeve crimped onto the end of a stranded wire before it goes into a screw terminal. He says to crimp it with the proper tool; tightening the terminal screw does not crimp it. The second warning is about passive crossovers, the networks of coils and capacitors that split the sound between woofer and tweeter. A "2nd-order" network cuts 12 dB per octave, and its tweeter section includes a coil wired across the tweeter (the "shunt coil"). If the tweeter is disconnected or blown, the amp sees that coil instead. At the network's resonant frequency, it then behaves almost like a dead short (a path with almost no resistance). So in practice: never play a 2nd-order or steeper passive network without its tweeter connected.

4.12 Inside factory systems: what factory amps and DSPs actually do

What this part is about: What the factory ("OEM" or "OE", meaning original equipment) audio system in a modern car is really doing behind the scenes. His view is that factory amplifiers and radios do a lot of careful sound processing, and you need to understand it before you replace or bypass anything.

4.12 #1 A premium factory amp is a powerful DSP tuned by acousticians; swapping it for a big amp with no EQ is a downgrade.

> For about ten years he used Harman OE amps as the DSPs in R&D cars, and his 2012 car ran OE Hyundai amps for EQ, crossover, delay and limiting. Bose, Panasonic and Harman engineers (many with PhDs) design these systems; calling them junk for lacking bass, then rearranging the car to install "the thing we've always installed", is "a big mistake". RARE
>
> **Sources:** DIYMA 2008-07-09 · DIYMA 2012-04-13 · CMA Connected Oct 2021 @37:25

In plain English: A "premium" factory system is the upgraded, often brand-name sound system (Bose, Harman Kardon and so on). Its amplifier is not just an amplifier: it contains a DSP (digital sound processor) that adjusts the sound, tuned by acousticians (sound engineers). His evidence: for about ten years he used Harman factory amps as the processors in research cars, and the car he ran in 2012 used factory Hyundai amps to do its EQ (tone shaping), crossovers (splitting sound between speakers), delay and limiting. He says engineers at Bose, Panasonic and Harman, many with PhDs, design these systems. Calling them junk because they lack bass, then tearing up the car to install "the thing we've always installed", is in his words "a big mistake". So replacing that amp with a big amp that has no EQ is a step down. Many people believe factory amps are weak junk to rip out first.

4.12 #2 Almost every factory system with an outboard amp applies EQ, crossovers and delay; head units now add all-pass filters and limiting too.

> OEMs often delay the rears for ambience and decorrelation; head units add all-pass filters for the other seat, EQ, and compression to protect small speakers from bass. Identify what's there before assigning a cause. RARE
>
> **Sources:** DIYMA 2013-12-17 · DIYMA 2021-02-03

In plain English: An "outboard" amp is a separate amplifier box rather than one built into the radio. He says almost every factory system with one applies EQ (tone shaping), crossovers (splitting frequencies between speakers) and delay (holding some speakers back by a few thousandths of a second). Carmakers often delay the rear speakers to create ambience and decorrelation, which means making the rear sound less identical to the front. Newer radios also add all-pass filters, which shift the timing of sound around a chosen frequency without changing its loudness; these are there for the other front seat. They also add EQ and compression (automatically turning down loud parts) to keep bass from damaging small speakers. So in practice: if something sounds odd, first find out what processing is already there before deciding what caused it.

4.12 #3 Basic systems process in the head unit's audio-controller chip.

> Programmable bass and treble, often a simple 3-band parametric, usually stereo-linked (one set front, one rear). Radios lack real EQ because nobody writes it into the spec: 31 bands instead of 5 is "a few lines of copy/paste" plus UI work.
>
> **Sources:** DIYMA 2013-03-17 · DIYMA 2016-06-03

In plain English: In basic factory systems with no separate amp, the sound adjustments happen inside one chip in the head unit (the radio), the audio-controller chip. He lists what it usually offers: programmable bass and treble, and often a simple 3-band parametric EQ (an equalizer whose bands can be moved to chosen frequencies). It is usually stereo-linked: left and right share

settings, with one set for the front and one for the rear. Why don't radios have a proper equalizer? He says it is because nobody writes one into the product specification. Going from 5 bands to 31 is "a few lines of copy/paste" in the code, plus work on the on-screen controls.

4.12 #4 Most factory EQ corrects the cabin, not cheap speakers.

> He estimates only 10–20% compensates for the drivers. Interior designers "own" the cabin, so OEM placement problems get solved with EQ and all-pass filters. RARE
>
> **Sources:** CMA 2023 @11:00 · DIYMA 2013-03-16

In plain English: When a factory system applies EQ (boosting or cutting certain frequencies), what is it fixing? Many people believe it mainly covers up cheap speakers. He says most of it corrects the cabin, meaning the car's interior with its glass, seats and odd speaker locations. He estimates only 10–20% of it makes up for the drivers (speakers). His reason: interior designers "own" the cabin and decide where everything goes. So when speakers end up in awkward places, the audio engineers solve that with EQ and all-pass filters (filters that shift the timing of sound without changing its loudness) rather than by moving the speakers.

4.12 #5 Volume, tone, fader, chimes, Bluetooth and nav prompts often live in the factory amp, not the radio.

> Many OE head units send fixed-level audio (OEM JBL Synthesis: under 1 V), with volume and tone applied in the amp on CAN, J1850 or proprietary commands. Some cars also cut volume ~6 dB when they think a door is open, and a miswired door switch can trigger that with the door shut.
>
> **Sources:** DIYMA 2015-05-11 · DIYMA 2013-01-15 · DIYMA 2022-01-30

In plain English: In many cars, the radio is not where volume and tone are controlled. He says many factory head units send audio at a fixed level, whatever the volume knob says; the OEM JBL Synthesis system's output, for example, is under 1 volt. The radio then sends commands to the amplifier over the car's data network (CAN, J1850 or a maker's own system), and the amp applies volume, bass, treble and fader. Warning chimes, Bluetooth phone audio and navigation prompts can also live in the amp. He adds that some cars turn the radio down about 6 dB (a clearly noticeable drop) when they think a door is open, and a wrongly wired door switch can cause that even with the door shut. So in practice: the factory amp may be doing jobs you can't see.

4.12 #6 Some factory amps shut down with no speaker load, and factory noise cancellation can look like a fault.

> No-load protection guards the output filter if a speaker opens (JBL's MS-A5001 "High 2" input adds a thermistor load), so check an OE amp's output with and without a load. Bass that briefly "overwhelms" an integrated system can be the factory ANC at work. RARE
>
> **Sources:** DIYMA 2015-04-03 · DIYMA 2015-09-19

In plain English: Two factory behaviors that can fool you into thinking something is broken. First, some factory amps switch off if no speaker is connected. That "no-load protection" guards the amp's output filter if a speaker wire breaks or a speaker dies. JBL's MS-A5001 amplifier has a "High 2" input setting that adds a thermistor (a resistor whose value changes with heat) as a load for exactly this reason. So in practice: test a factory amp's output both with and without a load before

deciding it is faulty. Second, if bass seems to briefly "overwhelm" a system tied into the factory audio, he says it can be the factory ANC (active noise cancellation, which plays sound through the speakers to cancel engine and road drone) at work.

4.12 #7 Step-up factory systems aim for every seat to sound similar, and their drivers are often good.

> On average aftermarket systems sound worse than step-up OE (his rental cars vs shop demo cars); factory Harman drivers (e.g. BMW) beat many aftermarket speakers, and owners' complaints are about tuning, bass and Logic 7. Aftermarket listeners judge only bass and loudness and miss the envelopment and two-seat imaging. RARE
>
> **Sources:** DIYMA 2008-12-21 · DIYMA 2022-01-10 · Absolute Electronix 2023 @56:27

In plain English: A "step-up" factory system is the optional upgraded audio package. He says these aim to make every seat sound similar, and their speakers are often good. Comparing rental cars he has driven with shop demo cars, he finds that on average aftermarket systems (installed after purchase) sound worse than step-up factory ones. Factory Harman speakers, for example in BMWs, beat many aftermarket speakers; owners' complaints are about the tuning, the bass and Logic 7 (Harman's surround processing). His point is that aftermarket listeners judge only bass and loudness. They miss two things factory systems do well: envelopment (the feeling of being surrounded by sound) and imaging that works from both front seats. Many people believe any aftermarket system beats a factory one.

4.12 #8 Where factory systems are weak.

> He never heard a Harman OE center that didn't "suck"; HK/Mercedes 3" metal cones lose dust caps (fix with thick super glue; dealer parts ~$180); old Bose ran 4" drivers at ½ Ω with no tweeters; factory Logic 7 lacks power and bass. Factory high-level outputs clip near maximum volume.
>
> **Sources:** DIYMA 2016-01-18 · DIYMA 2015-07-30 · DIYMA 2018-05-22

In plain English: His list of weak spots. He never heard a center channel (a speaker in the middle of the dash) in a Harman factory system that didn't "suck". In Harman Kardon systems in Mercedes cars, the 3-inch (about 75 mm) metal-cone speakers lose their dust caps (the small dome in the middle of the cone); he says thick super glue fixes them, while dealer parts cost about $180. Old Bose systems ran 4-inch (about 100 mm) speakers at ½ ohm (a very low electrical load) with no tweeters (small treble speakers). Factory Logic 7 systems lack power and bass. And factory high-level outputs (the speaker-wire signals) clip, which means they distort because they run out of voltage, near maximum volume.

4.12 #9 Factory dash "middlers" aimed at the windshield work for two reasons.

> The narrowing HF beam reflects off the glass to everyone (a "poor man's tweeter"), and boundary gain (+6 dB per halving of space) lets a 2.5" dash speaker play to 200–250 Hz on the 15–20 W an OEM amp gives it. RARE
>
> **Sources:** CMA Connected Oct 2021 @32:43 · CMA Connected Oct 2021 @34:22

In plain English: "Middlers" are small midrange speakers in the top of the dashboard, often pointed up at the windshield. Many people assume that's just a cheap compromise. He says it works for two reasons. First, as the pitch rises, a speaker's sound narrows into a beam; that beam

bounces off the glass and reaches everyone, which he calls a "poor man's tweeter". Second, boundary gain: a speaker tucked against surfaces gets louder, +6 dB each time the space it plays into is halved. That lets a 2.5-inch (65 mm) dash speaker play down to 200–250 Hz on just the 15–20 watts a factory amp gives it. Hz (hertz) measures pitch: the lower the number, the deeper the sound.

4.13 Getting the factory signal: where to tap and what to measure first

What this part is about: If you keep the factory radio and add your own amps or processor, you must take ("tap") the sound signal from somewhere in the factory system. This part covers what kind of signal you'll find, where to tap it, and how to measure it first without damaging anything.

4.13 #1 First learn what kind of output you have: its voltage, and whether it is single-ended or floating (balanced, BTL).

> Speaker outputs are almost always ~8 V and look identical; "low level through a LOC is a terrible waste of output", and BMW "voltage problems" are usually an installer's LOC or a removed amp. Voltmeter test: each lead ~6 V DC to chassis means balanced; into a balanced input, + to hot, − to cold, ground left open. RARE
>
> **Sources:** DIYMA 2009-08-24 · DIYMA 2009-01-21

In plain English: Before connecting anything, find out what kind of output the factory system gives you. A "single-ended" output has one signal wire measured against ground (the car's metal), like an RCA plug. A "floating" output, also called balanced or BTL (bridge-tied load), uses two wires carrying opposite copies of the signal, and neither is tied to ground. He says factory speaker outputs are almost always about 8 volts, and you can't tell what kind you have just by looking at the wires. A LOC (line output converter) turns that speaker signal down to the low RCA level, and he calls doing so "a terrible waste of output". He adds that BMW "voltage problems" are usually caused by an installer's LOC or a factory amp that was removed. His test: with a voltmeter on DC volts, measure each speaker lead to the chassis (car body). About 6 V DC on each means the output is balanced. To connect it to a balanced input: + to hot, − to cold, and leave ground unconnected.

4.13 #2 Balanced speaker-level signals beat RCA pre-outs.

> They sit ~6 V off ground, are higher in voltage and lower in impedance, and reject noise just by twisting the pair; many factory systems already use them ("silly" that aftermarket radios don't). The catch is a head unit that clips at the top of its volume range; Helix high-level inputs take 7.1 V RMS. RARE
>
> **Sources:** DIYMA 2013-01-15 · DIYMA 2021-09-25 · DIYMA 2022-02-27

In plain English: "Pre-outs" are the low-voltage RCA outputs on an aftermarket radio, meant for feeding an amp. Many people believe these are always cleaner than speaker-level signals. He says balanced speaker-level signals (two wires carrying opposite copies of the signal) are better. They sit about 6 volts away from ground, carry a higher voltage, and have a lower impedance (roughly, they are "stiffer" and harder for noise to disturb). Simply twisting the two wires together makes them reject noise. (That works because noise picked up lands on both wires equally, and a balanced input cancels whatever the two wires share.) Many factory systems already use them,

and he calls it "silly" that aftermarket radios don't. The catch: some head units clip (distort) at the top of their volume range. Helix processors' high-level inputs take 7.1 volts RMS (a kind of average voltage).

4.13 #3 Where to tap: after the factory amp to keep chimes and controls; before it only when its processing can't be undone.

> MS-8 recipe: keep the factory amp, feed its outputs to speaker-level inputs, radio REM → DSP REM-in, DSP REM-out → factory amp. Toyota JBL Synthesis's fixed-level balanced head-unit output lets you drop the factory amp if the DSP has volume control. **Conflict:** a secondhand ClubLexus report says he found MS-8 worked in a Lexus GS only when tapped before the amp.
>
> **Sources:** DIYMA 2012-12-25 · DIYMA 2016-03-14 · ClubLexus (secondhand)

In plain English: Where should your new processor get its signal? He says take it after the factory amp, so you keep the chimes and controls the amp handles. Take it before the amp only when the amp's processing can't be undone. His recipe for the MS-8 (a JBL processor) is to keep the factory amp and feed its outputs into the MS-8's speaker-level inputs. For switching on, the radio's remote (turn-on) output goes to the DSP's remote input, and the DSP's remote output goes to the factory amp. One exception: Toyota's JBL Synthesis head unit puts out a fixed-level balanced signal, so you can remove the factory amp if your DSP has its own volume control. The [Conflict] tag: a secondhand report on the ClubLexus forum, written by someone else, says he found the MS-8 worked in a Lexus GS only when connected before the factory amp. So his general recipe and that car-specific report point different ways.

4.13 #4 Bypassing the factory amp so it won't "touch" the music is an arbitrary goal with costs.

> A box can't tell music voltage from phone voltage. With a fixed-level radio you lose volume control (add a DSP with volume or a pot) and possibly chimes, Bluetooth and nav prompts.
>
> **Sources:** DIYMA 2013-04-23 · DIYMA 2012-09-11

In plain English: Some installers want to bypass the factory amp so that it won't "touch" their music. He calls that an arbitrary goal, and one with costs. His point is that a box can't tell music voltage from phone voltage; to the electronics, a signal is just a voltage. If the radio sends a fixed-level signal (the same level whatever the volume knob says), bypassing the amp loses your volume control. You would then need a DSP with its own volume control, or a pot (a volume knob). You may also lose warning chimes, Bluetooth phone audio and navigation prompts, since those can live in the factory amp.

4.13 #5 Adding a sub: tap the factory amp's sub channel so the sub tracks the volume knob.

> Use an amp with speaker-level differential inputs or a passive LOC; a powered LOC with its own level control means re-balancing the sub at every volume change.
>
> **Sources:** DIYMA 2009-07-06

In plain English: To add a subwoofer (a big speaker for deep bass) to a factory system, he says to take the signal from the factory amp's own sub channel. That way the new sub gets louder and quieter with the volume knob, in step with the rest of the system. Connect it with an amp that has

speaker-level differential inputs, or with a passive LOC (a line output converter with no power supply of its own). He warns that a powered LOC with its own level control means re-balancing the sub against the other speakers every time you change the volume.

4.13 #6 Measure every factory output first for response, delay and polarity; a rough measurement beats none.

> Response: test tones on CD, a multimeter and his Excel sheet, or a sound card and REW. Delay: an impulse via Aux, or a CD with a pulse every ~15 s and a 2-channel scope program. Polarity: an Audacity DC-offset track and a voltmeter; compare each output with unaltered pink noise, not channel with channel. RARE
>
> **Sources:** DIYMA 2016-02-08 · DIYMA 2015-06-30 · DIYMA 2021-05-20

In plain English: Before tuning, he says to measure every factory output for three things. Frequency response is how loud each pitch comes out. Delay is whether some outputs are held back in time. Polarity is whether the signal pushes the speaker out or pulls it in. Even a rough measurement beats none. For response, play test tones from a CD and read the voltage with a multimeter into his Excel sheet, or use a computer sound card and REW (Room EQ Wizard, free measuring software). For delay, play an impulse (a sharp click) through the Aux input, or a CD with a pulse every ~15 seconds, and view it with a 2-channel oscilloscope program. For polarity, make a DC-offset track (a steady one-direction push) in Audacity (a free audio editor) and check the voltage's sign with a voltmeter. Compare each output with the original, unaltered pink noise (a test hiss containing all pitches). Comparing one channel with the other, he says, can't show which one carries the filter.

4.13 #7 Never connect both radio channels to one computer input.

> Headset-adapter leads tie one side of each speaker output to computer ground, so joining both channels ties their negatives and can trip the radio's output protection. Measure one channel at a time, put a LOC in front, or use a USB interface with balanced XLR inputs (hot and cold, ground open) plus an attenuator. RARE
>
> **Sources:** DIYMA 2021-11-01 · DIYMA 2021-11-01

In plain English: A common cheap way to measure a radio's output is a headset adapter that plugs into a computer. He warns that these adapters connect one side of each speaker output to the computer's ground. If you connect both the left and right radio channels this way, their negative wires get joined, and that can trip the radio's output protection so it shuts off. His safe options: measure one channel at a time; put a LOC (line output converter) in front of the adapter; or use a USB audio interface with balanced XLR inputs (a three-pin professional connector). Wire your test leads to its "hot" and "cold" pins, leave ground unconnected, and add an attenuator, a part that lowers the voltage so the interface isn't overloaded.

4.13 #8 Audiofrog amps take factory outputs directly (product-specific).

> Speaker-level inputs to 30 V peak at 1.2 kΩ; line level 0.5–10 V peak; turn-on by signal sense, 6 V BTL offset detection or remote wire. The A600.1D sums its two inputs, and the A150.4D's 3-channel mode feeds outputs 3–4 the sum of inputs 1–2.
>
> **Sources:** Audiofrog amplifiers · A-Series manual

In plain English: This point is specific to Audiofrog's own amplifiers (the company he co-founded). They can take factory outputs directly. Their speaker-level inputs accept up to 30 volts peak and present 1.2 kΩ (1,200 ohms, a light load) to the factory system. Their line-level (RCA) inputs accept 0.5 to 10 volts peak. They can switch on three ways: by sensing a signal, by detecting the 6-volt DC offset that a balanced (BTL) factory output sits at, or by a remote (turn-on) wire. The A600.1D, a one-channel amp, combines ("sums") its two inputs. In 3-channel mode, the A150.4D feeds outputs 3–4 the combined signal of inputs 1–2.

4.14 Factory all-pass filters, summing, de-EQ and factory delay

What this part is about: Factory systems often contain hidden timing tricks: all-pass filters and delays. This part explains what they are, how to detect them, what happens when you try to undo them or combine ("sum") factory channels, and how his own approach to this changed over the years.

4.14 #1 Factory all-pass filters let both front seats image, because delay works for only one seat.

> A 1st-order all-pass is 90° at its frequency (0° above, 180° below); a 2nd-order is 180° (≈360° below). Typical layouts: one channel polarity-reversed plus a 1st-order all-pass restoring bass phase, centering vocals around 250 Hz; or two 1st-order all-passes near 250 and 500 Hz; Bose often stacks several on fronts and rears. RARE
>
> **Sources:** DIYMA 2021-05-13 · DIYMA 2021-05-16 · DIYMA 2021-05-19

In plain English: In a car you sit closer to one side's speakers, so their sound arrives first and pulls the stereo "image" (where the singer seems to stand) toward that side. Aftermarket tuners fix this with delay, holding back the nearer speakers a few thousandths of a second. He points out that delay works for only one seat. Factory systems instead use all-pass filters so both front seats get a usable image. An all-pass filter leaves loudness alone and changes only phase, the timing of a wave measured in degrees (360° is one full cycle). A 1st-order all-pass shifts phase by 90° at its chosen frequency, about 0° well above it and 180° well below. A 2nd-order one shifts 180° at its frequency (about 360° below). The typical factory layouts he describes: one channel's polarity reversed, plus a 1st-order all-pass that brings the bass back in step, centering vocals around 250 Hz; or two 1st-order all-passes near 250 and 500 Hz. Bose often stacks several on front and rear speakers. Many people believe delay is the only way to fix imaging in a car.

4.14 #2 Two tests to tell factory all-pass filters from factory delay.

> By ear: with all-pass filters a centered vocal is centered (not pinpoint) from the driver's seat and both sides are audible from the passenger seat; with delay, everything collapses into the passenger door. On an analyzer: mono pink noise, view L+R; one big dip at 200–500 Hz means all-pass, a dip followed by evenly spaced dips (a comb) means delay. RARE
>
> **Sources:** DIYMA 2021-10-26

In plain English: How can you tell which trick a factory system uses? His listening test: play a song with a centered vocal. With all-pass filters, from the driver's seat the vocal sounds centered, though not pinpoint sharp, and from the passenger seat you can still hear both sides. With delay, from the passenger seat everything collapses into the passenger door. His analyzer test: play mono pink noise (a test hiss with the same signal in both channels) and view the left-plus-right sum on

an analyzer, a screen that shows loudness at each pitch. One big dip between 200 and 500 Hz means all-pass filters. A dip followed by evenly spaced dips, a pattern called a "comb", means delay. (A comb appears when a sound is mixed with a time-shifted copy of itself.)

4.14 #3 "Reversed" OEM polarity is usually deliberate, so test empirically.

> Factory harnesses built thousands of times can't be miswired; if it sounds better as connected, it was designed around the phase EQ. **Conflict:** 2015, a 1st-order all-pass flips polarity below its corner; 2021, pulse checkers read all-passes like crossovers, so they don't show a reversal. **RARE**
>
> **Sources:** DIYMA 2021-05-20 · DIYMA 2015-12-07 · DIYMA 2021-05-16

In plain English: Polarity is whether a speaker pushes out or pulls in for the same signal. When people find a factory speaker that seems wired "backwards", they often assume it's a mistake. He says it's usually deliberate: factory wiring harnesses are built thousands of times and can't be miswired. If it sounds better as connected, it was designed to work with the factory's phase EQ (its all-pass filters). So in practice: judge by testing and listening, not by assuming an error. The [Conflict] tag notes two statements of his. In 2015 he said a 1st-order all-pass flips polarity below its corner frequency (the frequency it is set to). In 2021 he said the pulse-type polarity checkers people use read all-pass filters the same way they read crossovers, so they don't show an all-pass as a reversal.

4.14 #4 With factory all-pass filters in place, don't add delays.

> They are a different fix for the same problem and take the place of delay; delay on top makes them work poorly, and the passenger is spared a driver-seat tune's "nightmare". An all-pass can't be inverted: match it with the same filter on the other channel, or use FIR at the cost of latency (hands-free calls suffer). Factory Subaru case: leave polarity alone, no front time alignment. **RARE**
>
> **Sources:** DIYMA 2021-05-27 · DIYMA 2021-03-26 · DIYMA 2021-05-11

In plain English: If the factory system already has all-pass filters, he says don't add delays. Both are fixes for the same problem (imaging in a car where you sit off-center), and the all-pass filters take the place of delay. Adding delay on top makes them work poorly. The upside is that the passenger is spared the "nightmare" a driver's-seat-only tune sounds like from the other seat. You also can't simply undo an all-pass filter, because no filter exactly reverses it. Instead, put the same filter on the other channel so the two match. Or use FIR, a type of digital filter that can reshape timing, at the cost of latency (extra delay through the system), which makes hands-free phone calls suffer. In a factory Subaru case, he advised leaving polarity as it was and adding no front time alignment (delay).

4.14 #5 Finding all-pass filters.

> In REW they're easy to see with flat response and the initial delay removed (T=0). Or record the radio's L and R outputs and mix them to mono in a DAW: phase differences show up as response differences; DSSF3 (ymec.com) shows correlation directly.
>
> **Sources:** DIYMA 2021-05-16 · DIYMA 2021-05-16

In plain English: Two ways to spot all-pass filters, according to him. In REW (Room EQ Wizard, free measuring software), they're easy to see when the response is flat and the starting delay is removed, by setting T=0 so the time it takes the signal to arrive doesn't hide the phase pattern. Or

record the radio's left and right outputs on a computer and mix them to mono in a DAW (digital audio workstation, recording software). Wherever the two channels' timing differs, the mono mix shows a change in loudness. He also mentions DSSF3 software (from ymec.com), which shows correlation (how alike the two channels are) directly.

4.14 #6 All-pass filters were the one thing MS-8 handled badly.

> A phase-inverting midrange all-pass in a Lexus factory amp (2010) and Bose's phase-reversing all-passes in center-less systems (2016) defeated its summing; the upside, a 2012 Fiat 500 Bose system images in both front seats with no center.
>
> **Sources:** DIYMA 2010-08-05 · DIYMA 2016-05-11

In plain English: This point is about the MS-8, a JBL processor from a specific period, built to take in factory signals and combine ("sum") them back into plain stereo. He says all-pass filters were the one thing it handled badly. Two examples: in 2010, a Lexus factory amp with an all-pass filter on the midrange that inverted its phase; and in 2016, Bose systems without a center speaker that used phase-reversing all-pass filters. Both defeated the MS-8's summing. The upside of those filters: a 2012 Fiat 500 with a Bose system images (gives a clear stereo picture) in both front seats without any center speaker.

4.14 #7 Summing factory outputs through LOCs is "the worst way", because factory crossovers were designed to sum acoustically.

> Summed electrically, their phase shifts turn asymmetrical and leave holes (e.g. a midbass hole when factory sub and front channels are summed). His "correct" method: processor 1 (e.g. a PS8) EQs and delays each factory channel to sum through symmetrical resistors, processor 2 tunes; ~$2,000, or one speaker-level-input DSP if there is only a factory radio. RARE
>
> **Sources:** DIYMA 2015-06-13

In plain English: "Summing" means combining several factory outputs (for example tweeter, midrange and sub channels) back into one full-range signal for your own processor. He calls doing this with LOCs (line output converters) "the worst way". Factory crossovers were designed to blend acoustically, in the air, from speakers in their places. Wired together electrically, their phase shifts turn asymmetrical, as he puts it, and leave holes in the sound. One example is a midbass hole when the factory sub and front channels are summed. His "correct" method uses two processors. Processor 1 (for example a PS8) applies EQ and delay to each factory channel so they combine properly through symmetrical resistors (a simple resistor network that mixes them), and processor 2 does the tuning. That costs about $2,000, or a single DSP with speaker-level inputs if the car has only a factory radio. Point 11 below explains how his approach later shifted.

4.14 #8 Uncorrected factory delay is "an imaging nightmare", and few integration DSPs remove it.

> It comb-filters and makes your own time alignment wrong. In 2013 almost no de-EQ DSPs undid delay and in 2015 only the discontinued MS-8 did; AudioControl's 2016 DSP could delay inputs before summing. Choose an integration DSP by its input-correction tools. RARE
>
> **Sources:** DIYMA 2013-12-17 · DIYMA 2015-05-10 · DIYMA 2016-05-01

In plain English: Some factory systems delay certain speakers. If that delay is left in when you take the signal for your own processor, he calls it "an imaging nightmare": it causes comb filtering (evenly spaced cancellations in the sound), and it makes your own time alignment (delay settings) wrong. An "integration DSP" is a processor made to take factory signals. He says few remove factory delay. In 2013 almost none of the "de-EQ" processors (ones that undo the factory's tone shaping) undid delay, and in 2015 only the MS-8, by then discontinued, did. AudioControl's 2016 DSP could delay its inputs before summing them. So in practice: choose an integration DSP by the tools it has for correcting its inputs.

4.14 #9 Reading a bad summed measurement (Audi/Bose case).

> A hole near 200 Hz = asymmetrical factory crossovers plus front delay vs the sub; ~5 kHz 20 dB hot = DSP level mismatch; narrow notches near 4.5 and 9 kHz = comb filtering from uncorrected tweeter/mid delay.
>
> **Sources:** DIYMA 2015-06-13

In plain English: Here he diagnoses a bad measurement from an Audi with a Bose system, where the factory channels had been summed. A measurement graph shows loudness at each pitch. He reads three problems from it. A hole near 200 Hz comes from asymmetrical factory crossovers, plus the front speakers' delay relative to the sub. A region around 5 kHz that is 20 dB too loud (a large difference) comes from a level mismatch in the DSP. Narrow notches near 4.5 and 9 kHz are comb filtering, evenly spaced cancellations, caused by the tweeter-to-midrange delay that hadn't been corrected.

4.14 #10 Leave factory rear outputs out of the sum; double processing is fine.

> Rears are often out of phase or delayed, so summing them cuts highs and the input EQ lifts treble and hiss. A factory DSP feeding an aftermarket DSP breaks no rule; his own car ran several A/D/A conversions. RARE
>
> **Sources:** DIYMA 2014-03-02 · DIYMA 2013-10-08

In plain English: He says to leave the factory rear-speaker outputs out when summing the factory signal. Rears are often out of phase (timing reversed) or delayed, so adding them in cancels some of the treble. The processor's input EQ then boosts the treble back up, and with it the hiss. His second point: running one processor after another is fine. A factory DSP feeding an aftermarket DSP breaks no rule, and his own car ran several A/D/A conversions (analog to digital and back again). Many people believe processing a signal twice, or converting it back and forth, ruins the sound.

4.14 #11 "Get back to zero" first, or work with the factory processing?

> In upmixing factory systems he now wires each factory output one-to-one to an amp channel, adds speakers and power, and just retunes EQ and levels: the factory already holds the delays and crossovers. Taught this way, installers call it the best thing they've built. RARE **Changed view:** 2013–2016 his correct method was de-summing, de-EQ and delay removal (points 7–8); 2021–2023 says all-pass filters serve a purpose, are very hard to delete, and "back to zero" isn't always efficient.
>
> **Sources:** DIYMA 2015-06-13 · CMA Connected Oct 2021 @38:54 · Absolute Electronix 2023 @58:34

In plain English: "Back to zero" means undoing all factory processing to get a plain, flat stereo signal before tuning. An "upmixing" factory system creates extra channels, such as center, sides and rears, from stereo. In these, he now wires each factory output one-to-one to its own amp channel, adds speakers and power, and simply retunes the EQ and levels. His reasoning is that the factory processing already holds the delays and crossovers. He says installers taught this way call it the best thing they've built. The [Changed view] tag describes two positions. In 2013–2016 his "correct" method was to un-sum, undo the EQ and remove the delay (points 7 and 8). In 2021–2023 he says the factory all-pass filters serve a purpose and are very hard to delete, and that getting "back to zero" isn't always efficient.

4.15 Keeping factory features, and OEM upmixers (QLS, Logic 7) and digital links

What this part is about: Many premium factory systems use an "upmixer", a processor that spreads ordinary stereo music across more speakers. This part covers why they exist, how to keep or remove them, their quirks, how to test them, and why aftermarket gear rarely has one. It ends with digital audio links.

4.15 #1 Factory upmixers exist to make stereo better, not for movies.

> Logic 7-type processors steer ambience to the sides and rear and plant a center image for each passenger; in Pro Logic II, Logic 7 and DTS Neural the rears carry only ambience, hence "the rears don't play". Every car he has owned for 20+ years ran an upmixer, including his ~2004 Mini Cooper competition car (18 speakers, Logic 7). RARE
>
> **Sources:** DIYMA 2015-12-29 · DIYMA 2021-09-08 · OFCAP 19 @1:18:00

In plain English: An upmixer takes a normal two-channel stereo recording and creates extra channels from it, such as center, sides and rears. Many people believe surround processing is only for movies. He says factory upmixers exist to make stereo music better. Logic 7-type processors steer ambience (the room echo and reverb in a recording) to the side and rear speakers, and they place a center image (where the singer seems to be) in front of each passenger. In Pro Logic II, Logic 7 and DTS Neural (three upmixer types), the rear speakers carry only ambience. That's why owners complain "the rears don't play". Every car he has owned for over 20 years used an upmixer, including his roughly 2004 Mini Cooper competition car (18 speakers, Logic 7).

4.15 #2 OE systems are deliberately tuned for both front seats, so build on that.

> Carmakers chose good sound in both seats over a driver-only delay tune; better passenger sound costs nothing if the driver isn't compromised. Some cars image well but won't play loud: keep the imaging and add level, starting from what can't be removed (the radio, often the amp). RARE
>
> **Sources:** DIYMA 2021-09-09 · OFCAP 19 @3:15:06 · OFCAP 19 @1:04:03

In plain English: He says carmakers deliberately tune their systems for both front seats. Aftermarket tuners usually use delay to optimize the sound for the driver's seat alone, and carmakers chose good sound in both seats over that. His view: better sound for the passenger costs nothing, as long as the driver's sound isn't compromised. Some factory cars image well (give

a clear stereo picture) but can't play loud. So in practice: keep the imaging and add level (the ability to play louder), starting from the parts you can't remove, which are the radio and often the factory amp.

4.15 #3 Harman QLS 3-D: use it properly or remove it entirely.

> He found it in Hyundai/Kia, Maserati and Lincoln; about seven filters split center, side and in-between content, with "In the audience" and "On the stage" setting the wrap. Its outputs may not sum back to stereo, so tapping one output won't do; removing it in a Genesis may need a MOST adapter. RARE
>
> **Sources:** EDUCAR tech tip · OFCAP 19 @1:19:00

In plain English: QLS 3-D is a Harman upmixer. He found it in Hyundai/Kia, Maserati and Lincoln cars. It uses about seven filters to split the music into center content, side content and the in-between. Two settings, "In the audience" and "On the stage", control how much the sound wraps around you. His advice: either use it as intended, or remove it entirely. Its outputs may not add back up to normal stereo, so taking the signal from just one output won't give you the whole music. Removing it in a Hyundai Genesis may need a MOST adapter; MOST is a fiber-optic digital network some cars use to carry audio.

4.15 #4 Mercedes HK Logic 7 has no full-range stereo signal to tap, and on Mark Levinson/Logic 7 cars "DO NOT TRY TO SUM this car back together".

> Options: a Mobridge DA1 into the DSP; remove the Logic 7 amp and have the dealer recode the radio as no-amp; a DSP with its own volume control fed before the factory amp; or one replacing fronts, center and sub while sides and rears stay factory. RARE
>
> **Sources:** DIYMA 2015-12-28 · DIYMA 2021-09-26

In plain English: In Mercedes cars with Harman Kardon Logic 7, he says no output carries a full-range stereo signal; every output is an already-processed piece. For Mark Levinson cars with Logic 7 (Mark Levinson is Lexus's premium system), he wrote in capitals: "DO NOT TRY TO SUM this car back together". In other words, don't try to recombine the pieces into stereo. The options he gives: a Mobridge DA1 interface feeding your DSP; removing the Logic 7 amp and having the dealer recode (reprogram) the radio as a no-amp system; a DSP with its own volume control, fed from before the factory amp; or a DSP that replaces the front speakers, center and sub while the side and rear speakers stay factory. Many people believe any factory system can be summed back to stereo.

4.15 #5 Matrix upmixers mis-steer some music and magnify source clipping.

> Logic 7 and PL2 compare channels thousands of times a second (in-phase content forward, out-of-phase rearward), so producers' out-of-phase width tricks go rearward and sides and rears switch on and off. A clipping source sounds terrible through Logic 7: turn the head unit down and the processor up. At CES, EMMA judges needed an hour to find a track that "broke" Logic 7. RARE
>
> **Sources:** EDUCAR tech tip · DIYMA 2015-07-17 · DIYMA 2018-04-06

In plain English: A "matrix" upmixer like Logic 7 or Pro Logic II (PL2) works out where to send sound by comparing the left and right channels thousands of times a second. Sound that is the same in both ("in phase") goes forward; sound that is opposite in the two channels ("out of phase")

goes to the rear. Recording producers sometimes use out-of-phase tricks to make sounds seem wider, so the upmixer wrongly sends them rearward, and the side and rear speakers switch on and off. He also says a clipping (distorting) source sounds terrible through Logic 7. His fix is to turn the head unit down and the processor up. He also notes that at CES (a big electronics show), EMMA competition judges needed an hour to find a track that "broke" Logic 7.

4.15 #6 Test signals for upmixers.

> Use uncorrelated pink noise only to tune a matrix upmixer, correlated (mono) noise for everything else. Mono noise through Logic 7/PL2 comes mostly from the center with L/R ~6 dB down; invert one channel and it comes from the rears. RARE
>
> **Sources:** DIYMA 2021-10-08 · DIYMA 2017-08-05

In plain English: Pink noise is a test hiss containing all pitches. "Correlated" (mono) pink noise is identical in left and right; "uncorrelated" noise is different random hiss in each. He says to use uncorrelated noise only when tuning a matrix upmixer, and correlated (mono) noise for everything else. He describes how mono noise behaves through Logic 7 or PL2: it comes mostly from the center speaker, with the left and right speakers about 6 dB quieter. If you invert one channel (flip its polarity), it comes from the rear speakers instead.

4.15 #7 Why aftermarket DSPs lack upmixers: product managers, not cost.

> Upmixer licenses are cheap (discrete surround isn't); someone must understand it, fight for it past finance, and explain it to users.
>
> **Sources:** DIYMA 2021-08-01 · DIYMA 2017-07-07

In plain English: Why don't aftermarket processors include upmixers? He says it isn't the cost: upmixer licenses are cheap, though licenses for discrete surround (true multichannel formats like movie soundtracks) aren't. The obstacle is product managers. Someone inside the company must understand upmixing, argue for it past the finance people, and then explain to customers how it works and how to use it.

4.15 #8 Digital links and phone audio.

> Optical never became standard because a digital feed bypasses the radio's volume, so every amp needs its own volume or a bus; floating outputs into differential inputs nearly match its noise immunity. A cheap decoder can be tapped at its I2S bus with leads a few inches long. Through a processor, "a 100% reliable solution is more important than a completely transparent one". No Andy statement on A2B was found.
>
> **Sources:** DIYMA 2013-11-22 · DIYMA 2015-08-11 · DIYMA 2010-06-13

In plain English: Optical links carry audio as light, in digital form. He says they never became standard in cars because a digital feed bypasses the radio's volume control. Every amp would then need its own volume control, or a data bus to receive volume commands. He adds that floating (balanced) outputs feeding differential inputs come close to matching optical's immunity to noise. For a cheap decoder box with poor analog outputs, you can take the signal from its I2S bus, the internal digital link between its chips, but only with leads a few inches long. On phone audio, a

2010 case of Bluetooth calls through the MS-8 processor involved processing delay. There his principle was "a 100% reliable solution is more important than a completely transparent one". The researchers found no statement from him about A2B (a newer automotive digital audio bus).

4.16 Staged upgrades and factory-fit products

What this part is about: How he suggests upgrading a factory system one step at a time, keeping the factory processing as long as it helps. Several points describe Audiofrog products (his company) designed to fit factory locations; those are product-specific.

4.16 #1 New speakers alone are a real upgrade on a factory-EQ'd system, and a DSP fixes most "voicing" complaints.

> In Bose and other premium systems that already EQ out the car, similar speakers in the same places keep the factory EQ doing useful work. Before swapping speakers for their sound, add a DSP. RARE
>
> **Sources:** CMA 2023 @11:00 · DIYMA 2022-01-14

In plain English: Many people believe new speakers can't help a factory system until you replace or bypass its processing. He disagrees. In Bose and other premium systems, the factory EQ (tone shaping) already corrects for the car's interior. If you fit similar speakers in the same places, that factory EQ keeps doing useful work, so new speakers alone are a real upgrade. "Voicing" is a speaker's built-in tonal character, and people often switch brands because they dislike it. His advice is that a DSP (digital sound processor) fixes most of those complaints. So in practice: before swapping speakers because of how they sound, add a DSP.

4.16 #2 His staged path keeps the factory EQ and crossover until the last step.

> (1) Better speakers on the factory amp and EQ; (2) more power; (3) sub, rears, amps and a DSP. On a 2023 Chevy truck with Bose the tiers ran about $2,000/$3,500/$4,500/$7,500; demo each stage with binaural recordings on headphones. Gary Biggs's IASCA BMW kept every factory driver and added amps, an MS-8 and a sub. RARE
>
> **Sources:** CMA 2023 @12:05 · OFCAP 19 @49:01 · DIYMA 2022-01-10

In plain English: His recommended upgrade order keeps the factory EQ and crossover (the circuit that splits sound between speakers) until the last step. Step 1: better speakers, still on the factory amp and EQ. Step 2: more power (amplifiers). Step 3: a subwoofer, rear speakers, amps and a DSP. On a 2023 Chevy truck with Bose, he priced the tiers at about $2,000, $3,500, $4,500 and $7,500; the sources list these as speakers, then amps, then a subwoofer, then a DSP tuned for a center image. To let customers hear each stage, demo it with binaural recordings (recordings made with microphones in a dummy head's ears) played on headphones. He also cites Gary Biggs's competition BMW in IASCA (a sound-quality contest), which kept every factory speaker and added amps, an MS-8 processor and a subwoofer.

4.16 #3 GS25: for factory systems whose active crossover and EQ make passive component sets "a big problem".

> Common in GM, Toyota and Mercedes systems; the GS25 is a 2.5" full-range (200 Hz–20 kHz) with no crossover and break-off tabs for factory dash spots, taking the 50–75 W factory dash speakers can't. Pair it with GS60/GS690 doors or add a GS10 for an active 3-way. RARE
>
> **Sources:** GS25 listing · CMA 2023 @9:01 · CMA Connected Oct 2021 @36:43

In plain English: In many GM, Toyota and Mercedes factory systems, the factory amp itself splits the sound between the dash and door speakers (an "active" crossover) and applies EQ. A typical aftermarket "component set" comes with its own passive crossover box, and in these systems he calls that "a big problem". The GS25 is Audiofrog's answer. It's a 2.5-inch (65 mm) full-range speaker covering 200 Hz to 20 kHz, with no crossover, and break-off tabs so it fits factory dash locations. It handles 50–75 watts, which factory dash speakers can't. Pair it with GS60 or GS690 door speakers, or add a GS10 tweeter to make an active 3-way system.

4.16 #4 G62 coax is voiced for off-axis factory locations; G60S is packaged for factory doors (product-specific).

> The G62's treble rises to offset off-axis loss in lower doors and decks, and its tweeter sits low for factory grilles (~85% go behind one). G60S: inline crossovers by each driver, a tweeter kit that comes apart for factory spots, a shallow basket.
>
> **Sources:** G62 page · CMA 2023 @5:05 · G60S listing

In plain English: These are Audiofrog products. A coaxial (coax) speaker has its tweeter mounted in front of the woofer, as one unit. Factory speakers often sit low in the doors or on the rear deck, pointed away from your ears ("off-axis"), and treble fades off-axis. So the G62's treble is made to rise, offsetting that loss. Its tweeter also sits low so it fits behind factory grilles; about 85% of installs go behind one. The G60S component set is built for factory doors. Its crossovers sit inline, right by each speaker. Its tweeter mounting kit comes apart to fit factory spots, and it has a shallow basket (frame).

4.16 #5 Direct-fit tweeters and adapters (product-specific).

> GB10/GS10 in 3D-printed housings for BMW, Tesla, Audi, Porsche, Range Rover, F-150, Golf, Silverado/Sierra Bose and several Toyotas; GB10OE high-pass 2.5 kHz and GS10OE 3.5 kHz, both 24 dB/oct; GB25OE/GB40OE adapters for factory dash mids. Check housing dimensions first.
>
> **Sources:** Direct Fit Tweeters and Adapters · OE Tweeters collection

In plain English: These Audiofrog products replace factory tweeters (small treble speakers) without custom work. The GB10 and GS10 tweeters come in 3D-printed housings shaped for specific cars: BMW, Tesla, Audi, Porsche, Range Rover, Ford F-150, VW Golf, Chevy Silverado/GMC Sierra with Bose, and several Toyotas. The GB10OE has a built-in high-pass filter (it blocks lower sounds) at 2.5 kHz, and the GS10OE at 3.5 kHz, both cutting 24 dB per octave (a steep cut). GB25OE and GB40OE adapters fit factory dash midrange locations. So in practice: check the housing dimensions for your car first.

4.16 #6 Volvo S60 (c. 2001–2009): keep the factory Dynaudio speakers and add amp, EQ and sub.

> Replacing them is "a waste of money"; feed a real amp from the radio's sub-1 V DIN preamp signal (which carries remote turn-on) and flatten the head unit's hidden EQ. "How do I know this? I have one." RARE
>
> **Sources:** DIYMA 2012-03-25

In plain English: This is his advice for a Volvo S60 from about 2001–2009. Keep the factory Dynaudio speakers; replacing them, he says, is "a waste of money". Instead add an amplifier, EQ and a subwoofer. Feed the new amp from the radio's DIN preamp output, a low-level signal of under 1 volt on a DIN-type connector, which also carries the remote turn-on signal. Then flatten the radio's hidden EQ, the tone shaping the radio applies without showing it. His credential: "How do I know this? I have one."

4.16 #7 Bass is still the aftermarket's way in.

> OEM suppliers can't get a 40 lb sub box and a 60 A amp approved, so the core upgrade stays a better front plus added bass.
>
> **Sources:** OFCAP 19 @58:06

In plain English: Why is there still a market for aftermarket car audio? He says it's bass. Carmakers' audio suppliers can't get a 40-pound (18 kg) subwoofer box and an amplifier drawing 60 amps approved for a production car. So the core aftermarket upgrade remains a better front stage (the front speakers) plus added bass.

4.17 MS-8-era integration tips (JBL MS-8, 2010–2016; posts to 2022)

What this part is about: Tips for one specific product, the JBL MS-8, a car sound processor with automatic tuning that he helped bring to market. These tips come mainly from 2010–2016, and he kept answering questions about it in posts up to 2022. They apply to that unit and its era.

4.17 #1 MS-8 was designed first for factory integration: put the factory sub on input 7 or 8.

> It covers 48 speaker configurations by rule; its de-EQ must see a HF channel before a LF one. With an aftermarket radio, feed fronts to inputs 1–2 and skip input calibration; if it's too quiet on low-level inputs, check whether it's tapped at the radio or the factory amp. Era: MS-8
>
> **Sources:** DIYMA 2011-06-09 · DIYMA 2011-10-18 · DIYMA 2021-12-28

In plain English: This is about the JBL MS-8 (an era-specific product). He says it was designed first of all to work with factory systems. When connecting a factory system, put the factory subwoofer's signal on MS-8 input 7 or 8. It supports 48 speaker layouts and works by built-in rules. Its "de-EQ", the step that undoes the factory's tone shaping, must see a high-frequency (treble) channel before a low-frequency (bass) one; the point doesn't spell out the details of that rule. With an aftermarket radio, connect its front outputs to inputs 1 and 2 and skip the input calibration step. If the MS-8 is too quiet on its low-level (RCA) inputs, check whether it is connected to the radio's output or the factory amp's output.

4.17 #2 "Signal None" vs "Signal Low".

> None = no signal, or one altered beyond frequency and delay (e.g. a sample-rate error); Low = too little voltage; chimes during calibration also trigger "none" (he cites Ford).
>
> **Sources:** DIYMA 2010-07-09 · DIYMA 2014-11-10

In plain English: These are two messages the MS-8 (an era-specific product) shows while checking its inputs against its setup disc. He explains them. "Signal None" means either there is no signal, or the signal has been changed in some way beyond its frequency balance and timing. One example is a sample-rate error, where digital audio plays at a slightly wrong rate. "Signal Low" just means too little voltage. He adds that warning chimes playing during calibration also trigger "none", citing Ford cars.

4.17 #3 Hiss after integration usually means an EQ is boosting treble.

> MS-8's input de-EQ, output EQ and 31-band can each lift highs; if hiss drops with processing off, raise the tweeter amp gain and recalibrate. With a radio that adds heavy bass boost and treble cut (2013 Ford Fusion), raise its treble, set MS-8 ~−6 and calibrate at the highest volume still reading "OK OK OK". RARE
>
> **Sources:** DIYMA 2015-07-10 · DIYMA 2013-12-05

In plain English: This is MS-8-specific (an era-specific product). Hiss is a steady background "shhh". He says that hiss after connecting to a factory system usually means an equalizer is boosting the treble. The MS-8 has three stages that can each raise the highs: its input de-EQ, its output EQ and its 31-band user EQ. If the hiss drops when processing is switched off, raise the gain on the tweeter amplifier and run the calibration again. A 2013 Ford Fusion radio, for example, adds heavy bass boost and cuts treble. There, he says to raise the radio's treble, set the MS-8 volume around −6, and calibrate at the highest radio volume that still reads "OK OK OK" on the MS-8 display.

4.17 #4 Why MS-8 results go bad: clipped sweeps, one reversed midbass, running out of boost.

> Sweep with MS-8 around −20 and the sub amp near 2 V (if you can feel the sub sweep, its gain is too high); "MS-8 absolutely hates one midbass out of phase". Expect lower level after auto-EQ, since it prefers cuts: "Boosting all the holes is a bad way to EQ."
>
> **Sources:** DIYMA 2010-08-05 · DIYMA 2013-01-23

In plain English: This is MS-8-specific (an era-specific product). The MS-8 tunes itself by playing test sweeps (tones gliding from low to high) and measuring them. He names three reasons results go bad. One is sweeps so loud they clip (distort). Another is one midbass speaker with reversed polarity; in his words, "MS-8 absolutely hates one midbass out of phase". The third is the EQ running out of boost. His settings: run the sweeps with the MS-8 volume around −20 and the sub amp's gain near the 2-volt setting. If you can feel the sub sweep, not just hear it, the sub gain is too high. Expect the system to play a bit quieter after auto-EQ, because the MS-8 prefers cutting peaks to boosting dips: "Boosting all the holes is a bad way to EQ."

4.17 #5 First-time settings, and the sub-level workaround.

> Amps ~2 V; MS-8 −30 for sweeps; sub/front 80 Hz 24 dB/oct, tweeter 3.5 kHz, sides 100 Hz, subsonic 20 Hz 12 dB/oct; JBL MS amps "Low", pot fully CCW, amp filters off. MS-8 sets sub level from the 50–80 Hz sweep before EQ and can over-boost lows (e.g. small sealed boxes): reverse the sub's polarity and set the sub amp's crossover highest, or run the front as a 3-way with front-low at 70 Hz.
>
> **Sources:** DIYMA 2012-12-29 · DIYMA 2018-04-08 · DIYMA 2015-07-28

In plain English: This is MS-8-specific (an era-specific product): his starting settings for a first-time setup. Set amp gains around 2 volts and the MS-8 volume at −30 for the test sweeps. Cross the sub and front at 80 Hz with a 24 dB-per-octave slope (a steep cutoff). Set the tweeter at 3.5 kHz, the side speakers at 100 Hz, and a subsonic filter (which removes rumble too low to hear) at 20 Hz, 12 dB per octave. On JBL MS amps, set the input switch to "Low", turn the gain pot (knob) fully counter-clockwise, and switch the amp's own filters off. He also explains a quirk. The MS-8 sets the sub's level from the 50–80 Hz part of the sweep before it applies EQ, and that can over-boost the low bass, for example with small sealed boxes. His workarounds: reverse the sub's polarity and set the sub amp's crossover to its highest setting, or run the front as a 3-way with the front-low crossover at 70 Hz.

4.17 #6 What MS-8's correction does in OEM-style multichannel systems.

> EQ corrects speaker locations (e.g. "front right"), not individual crossover channels; delays exist so one-seat tuning works through the upmixer and are barely used in "Front" mode. Check it with a multi-mic average; outside correction of sides/rears fails; no real custom firmware exists (encrypted binary, one display update). RARE
>
> **Sources:** DIYMA 2022-02-03 · DIYMA 2021-09-28 · DIYMA 2018-04-08

In plain English: This is about how the MS-8 (an era-specific product) works in factory-style systems with many speakers. Its EQ corrects each speaker location, such as "front right", not each separate crossover channel, such as that location's tweeter and woofer on their own. Its delays are there so tuning for one seat still works through the upmixer (the processor that spreads stereo across more speakers), and they are barely used in its "Front" mode. To check its work, he says use an average of several microphone positions, not a single mic. Trying to correct its side and rear outputs with another processor fails. And there's no real custom firmware for it: its software is encrypted, and only one display update was ever released.

4.17 #7 Factory layouts with MS-8: bi-amp the front before adding discrete rears.

> For a 10-speaker OEM JBL 4Runner, parallel the D-pillar speakers with the sides and run 5.1 with the front bi-amped; with BMW under-seat systems he asked owners to try feeding the factory amp's sub output.
>
> **Sources:** DIYMA 2012-09-03 · DIYMA 2011-11-08

In plain English: This is MS-8-specific (an era-specific product). "Bi-amping" means giving each front speaker in a pair, the woofer and the tweeter, its own amplifier channel. "Discrete" rears are rear speakers with their own separate channels. His priority: bi-amp the front before adding discrete rears. For a Toyota 4Runner with the 10-speaker factory JBL system, he suggests wiring the D-pillar speakers (in the rearmost pillars) together with the side speakers, and running the

MS-8 in 5.1 mode with the front bi-amped. In his words, "There's a bigger benefit to biamping the front than in discrete rear channels." For BMWs with woofers under the seats, he asked owners to try feeding the factory amp's sub output to the MS-8.

4.17 #8 MS-8 hardware quirks.

> It won't boot in the cold because it waits for the slow LCD (disconnect the display), yet it can't run without its own display, which holds its own processor; random balance shifts usually mean an intermittent input.
>
> **Sources:** DIYMA 2022-01-05 · DIYMA 2019-01-13 · DIYMA 2018-12-15

In plain English: These are quirks of the MS-8 (an era-specific product). In cold weather it may not start, because on start-up it waits for its screen, and LCD screens respond very slowly when cold. Disconnecting the display gets around this. Yet the MS-8 can't run without its own display, because that display holds its own processor and software, so another screen can't stand in for it. The sources don't say more about how the cold-start workaround fits with this. If the sound randomly shifts to one side, he says that usually means an intermittent input, a connection that is cutting in and out.

4.18 Top rarely-discussed points in this section

- Cabin gain belongs to the car and every box type gets it, so build the most efficient box you can fit and EQ the peak (4.1 #1, #2, #6).
- Open windows turn the cabin into a vented box; convertibles get little cabin gain and want a bandpass box firing into the cabin (4.1 #9, #10).
- Bass gives away its location through distortion and rattles, not arrival time (4.2 #2).
- Cross the sub at 60 Hz and the midbass at 80 Hz (LR4), and lower the sub low-pass half an octave per 12 dB of sub boost (4.2 #6).
- "Half a wavelength of delay plus a polarity flip" fixes phase at one frequency only, and impulse auto-delay fails on subs: use a tape measure (4.2 #11–#13).
- Vented boxes tuned for minimum excursion beat small sealed boxes on huge amps, and box Q is immaterial when the sub is crossed below Fc (4.3 #1, #6, #7).
- A down-firing sub box is an accidental bandpass enclosure (4.3 #17).
- Behind the rear seat is the worst sub location; confirm a cross-car mode with a ~7 ft tape-measure path difference (4.5 #2, #3).
- Never port a door box into the door; with a sub, a high-passed free-air midbass is enough (4.6 #5).
- Music averages ~1/32 of its peak power, and a tweeter sees ~0.16 W of a "20 W" calibration (4.7 #1, #2).
- About 70% of GB25 warranty claims are DSP input clipping; set the DSP input for maximum volume at the source's loudest frequency (4.7 #6, 4.8 #1–#2).
- His 8-step gain method uses a sine wave on the sub amp only (4.8 #4).
- DSP amplifiers can't recover output lost to EQ cuts; high-end systems need amps with analog gain (4.8 #9).

- Hiss is not a ground loop, and differential inputs ended the ground-loop era; use twisted pair, not coax, into them (4.10 #1, #2, #5).
- Crimp, don't solder (Big 3 included); the ground wire may be thinner than the power wire; every added fuse is a resistor (4.11 #1, #3, #5).
- Never connect both factory radio channels to one computer input when measuring (4.13 #7).
- Tell factory all-pass filters from delay by ear or by mono-noise L+R, and don't stack delays on them (4.14 #2, #4).
- In upmixing factory systems, wire factory outputs one-to-one to new amps and keep the factory processing (4.14 #11).

Changed views and conflicts in this section

Changed views

- **Sub/midbass crossover:** ~80 Hz LR4 on both sides (2010–2015) → sub 60 Hz / midbass 80 Hz stagger because the sub gets boosted (2020–2022) (4.2 #6).
- **Sub delay:** "don't time-align subs" (2009–2010) → delay everything to the sub (2015) → tape-measure the sub like any speaker (2018–2023 PDFs, 2021) (4.2 #12).
- **All-pass filters on sub/midbass:** never made them work (2010–2013) → used them in his own car (2015–16) → fix the crossover instead, and use the high-pass side if you must (2021) (4.2 #15).
- **Gain overlap:** clip on a −10 dB tone, ~9–10 dB overlap (2009–2015) → "never appropriate" at the DSP input; −10 dB tones license tweeter-killing clipping (2021) (4.8 #5).
- **Gain-setting method:** scope and −10 dB sine (2008) → by ear (2014–2016) → DSP-era method with a sine on the sub only, the rest by pink noise and RTA (2018–2022) (4.8 #4).
- **Midbass Q (partial):** car midbass should be low-Q (2015; 2021 blog) → crossed above resonance its Q "is not really important" (2021 DIYMA) (4.6 #3).
- **Factory integration:** sum, de-EQ and remove factory delay with two processors (2013–2016) → keep the factory all-pass filters and processing, go one-to-one (2021–2023) (4.14 #11).

Conflicts

- **Impact:** midrange/3 kHz (2009, 2021 DIYMA) vs midbass–sub phase for club-style impact (CMA Connected Oct 2021) (4.2 #3).
- **Efficiency vs power:** "efficiency matters more" (2021) vs "Efficiency is overrated. Power is cheap." (2013, in a level-matching context) (4.7 #11).
- **Class-D blind demo:** ~800 demos/3 listeners (2022) vs ~1,500 demos/4 listeners (2023) (4.9 #3).
- **Xmax method:** one-way overhang, no fudge (2008) vs (coil − gap)/2 + 15% (2020) (4.3 #21).
- **1st-order all-pass and polarity:** flips polarity below its corner (2015) vs pulse checkers see no reversal (2021) (4.14 #3).
- **Lexus tap point:** secondhand ClubLexus report (tap before the amp) vs his general advice to tap after it (4.13 #3).

- **DSP starting level:** −12 dB in the title-attributed DIYMA guide vs −6 dB in the 2022 eight-step post (4.8 #4).

Gaps: no statement from Andy on A2B was found. On the Big 3 he only says to crimp it, not solder it, and lists it among noise "fixes" that don't help (4.10 #2, 4.11 #3). On stiffening caps he says only that a 2-farad cap is unnecessary at 1,000 W (4.11 #7).

SECTION 5

5. Myths, judging gear, buying and the industry

This section gathers what Andy Wehmeyer says about car-audio folklore, how to judge gear and the claims made for it, how to plan and buy a system and pick a shop, how the business and SQ competitions work, and how we listen to recordings. Points are merged from his DIYMobileAudio (DIYMA) posts (2008–2022), the Audiofrog Tech Blog, podcasts, livestreams and press; installation, tuning and bass detail lives in sections 2–4 and appears here only as short myth entries. Jargon used throughout: DSP = digital signal processor; LR4 = 24 dB/octave Linkwitz-Riley crossover; all-pass filter (APF) = a filter that shifts phase without changing level; Fs = a driver's free-air resonance; "B1"/"B2" = the Absolute Electronix (2023) and CMA Connected (2021) livestreams.

What this section is about: This is the myth-busting and real-world part of the guide. It collects things many car-audio fans believe that Andy Wehmeyer says are wrong or overstated, plus his advice on judging gear and the claims made for it, planning and buying a system, choosing a shop, the business side, sound-quality (SQ) competitions and listening to recordings. The jargon list at the end of the paragraph is a mini-dictionary. A DSP is a small computer that adjusts the sound. A crossover splits the music into bass, middle and treble so each speaker gets the part it plays best. Phase is the timing of a sound wave's ups and downs. Fs is the note a speaker naturally wants to vibrate at when it hangs in open air. B1 and B2 are just short names for two livestreams he appeared on.

5.1 Myths Andy debunks (Myth → What Andy says)

What this part is about: Each item names a popular car-audio belief (the "myth") and then, after the arrow, gives Andy's answer. The items are grouped by topic, such as digital audio, cables and power, amplifiers, and speakers. Items #1 to #38 come first; #39 to #76 follow.

DIGITAL AUDIO, FILES AND SOURCES

5.1 #1 Myth: hi-res files beat CD → They add data, not audible information.

> 44.1 kHz captures everything to about 22 kHz; there are no "stair steps", because after the anti-alias filter only one wave fits the samples, and upsampling adds nothing (his image: 8 oz of juice poured into a 16 oz glass). A hi-res release is often just a better master; he hears no benefit above CD and demos 320 kbps MP3s burned to CD. RARE
>
> **Sources:** Audiofrog blog Dec 2020 · DIYMA 2021-08-10 · DIYMA 2021-08-10b

In plain English: Many people believe "hi-res" music files, which store more samples per second and more bits per sample, sound better than a CD. A CD stores sound as 44,100 snapshots (samples) per second, written 44.1 kHz. He says that captures everything up to about 22 kHz, which is above the roughly 20 kHz top of human hearing. People often picture digital audio as a jagged staircase. He says there are no stair steps: after the anti-alias filter (a filter that removes anything too high-pitched for the sample rate), only one smooth wave can pass through all the samples, so

that wave is what comes back out. Upsampling, meaning converting a file to a higher sample rate, adds nothing new. His image: pour 8 oz of juice into a 16 oz glass and you still have only 8 oz of juice. When a hi-res release does sound better, it is often just a better master (a better-prepared final version of the recording). He hears no benefit above CD quality, and for his demos he plays 320 kbps MP3s burned to CD.

5.1 #2 Myth: a DSP with more bits or a higher sample rate sounds better → Pick a DSP for its tools.

> 16 bits already gives about 96 dB signal-to-noise ("like no noise"), and he endorsed 48 kHz/16-bit as enough. What matters is input ranges that match your source, LR4 slopes, fine delay steps and a parametric EQ you can actually use. RARE
>
> **Sources:** Audiofrog blog Dec 2020 · DIYMA 2016-05-23 · DIYMA 2021-08-09

In plain English: Many people shop for a DSP by its numbers: more bits, or a higher sample rate. Bit depth sets how far the music sits above the background noise of the digital system. He says 16 bits already gives about 96 dB between the loudest signal and that noise, which he describes as "like no noise", and he endorsed 48 kHz/16-bit as enough. So he says to pick a DSP for its tools instead: input ranges that match the voltage your source puts out, LR4 crossover slopes, fine steps in the delay setting, and a parametric EQ you can actually use. A parametric EQ lets you choose the exact frequency, width and amount of each boost or cut.

5.1 #3 Myth: MP3 and Bluetooth wreck sound quality → 256k is hard to tell from CD; 320k he can't tell at all.

> Perceptual coding drops sounds that louder sounds mask, and data compression doesn't change dynamic range. Bluetooth realistically carries about 320 kbps, so there is no CD-quality Bluetooth; ask whether your experience is diminished, not whether a difference is detectable. RARE
>
> **Sources:** DIYMA 2013-01-19 · SQOLOGY Pt 1 @27:05 · DIYMA 2021-08-09

In plain English: Many people believe compressed formats like MP3, and Bluetooth, wreck sound quality. MP3 uses perceptual coding: it drops sounds you could not hear anyway, because a louder sound at nearly the same time and pitch masks (covers) them. Bitrate, in kbps (thousands of bits per second), is how much data is kept. He says 256 kbps is hard to tell from a CD, and he can't tell 320 kbps from a CD at all. Data compression (making the file smaller) is also not the same thing as squashing loud and soft parts together, so it doesn't change dynamic range. Bluetooth realistically carries about 320 kbps, so he says there is no CD-quality Bluetooth. His advice is to ask whether your listening experience is actually diminished, not whether a difference can be detected at all.

5.1 #4 Myth: the file or stream is the weak link → The playback system and its EQ matter far more.

> Bad EQ degrades music more than any format conversion; FLAC into a poor speaker "won't fix that", and most bad systems fail on basics (distortion, missing bandwidth, rough response). RARE
>
> **Sources:** DIYMA 2014-07-11 · SQOLOGY Pt 1 @39:06

In plain English: Many people believe the music file or streaming service is the weak link, so they hunt for the best format. He says the playback system and its EQ matter far more. Bad EQ (equaliser settings that boost or cut the wrong frequencies) degrades music more than any format conversion does. Playing FLAC, a lossless full-quality format, through a poor speaker "won't fix

that". He says most bad systems fail on basics: distortion, missing bandwidth (part of the range from deep bass to high treble isn't reproduced), and a rough frequency response (some notes much louder or quieter than others).

5.1 #5 Myth: vinyl (or the hi-res version) sounds better → Not by any objective measure; you're usually hearing a different mix.

> Vinyl and hi-res releases often get a different master with less compression, so compare formats only from the same source (rip the same record to each). Mono bass was a vinyl limit (keeping the needle in the groove), not a rule. RARE
>
> **Sources:** DIYMA 2021-08-10 · DIYMA 2021-08-20

In plain English: Many people believe vinyl records, or the hi-res version of an album, sound better than a CD or a normal file. He says not by any objective measure: you are usually hearing a different mix. Vinyl and hi-res releases are often given a different master (the final prepared version of the recording) with less compression, meaning loud and soft parts are squeezed together less. So he says to compare formats only from the same source, for example by ripping (recording) the same record into each format. He adds that keeping bass in mono (the same in both channels) was a limit of vinyl, needed to keep the needle in the groove, not a rule of good sound.

5.1 #6 Myth: every extra A/D–D/A conversion or digital volume control ruins the sound → "Wildly overblown".

> His own car converted 3–4 times with digital volume first, and no listener heard a problem; digital volume loses resolution only in extreme cases (1000 W channels played very quietly). RARE
>
> **Sources:** DIYMA 2012-04-13 · DIYMA 2013-10-08

In plain English: An A/D conversion turns sound into numbers, and a D/A conversion turns the numbers back into sound. Many people believe each extra conversion, or a volume control that works digitally, ruins the sound. He calls this "wildly overblown". His own car converted the signal 3 to 4 times, with digital volume control first in the chain, and no listener heard a problem. He says digital volume loses resolution (fine detail) only in extreme cases, such as 1000 W channels played very quietly.

5.1 #7 Myth: content above 20 kHz matters → Its audibility is "highly disputed" and he has never heard it.

> Tweeters extend to 40 kHz only to push breakup peaks out of the audible band, so a "40 kHz" spec is marketing. RARE
>
> **Sources:** DIYMA 2016-01-16 · DIYMA 2015-04-12

In plain English: Human hearing is usually said to top out around 20 kHz (20,000 vibrations per second). Many people believe music content above that matters. He says its audibility is "highly disputed" and that he has never heard it. Some tweeters (small speakers for high notes) are specified up to 40 kHz. He explains they reach that high only so their breakup peaks land above the range we can hear. Breakup is when a tweeter's dome stops moving as one solid piece, which creates sharp peaks in its output. So, in his view, a "40 kHz" spec is marketing.

5.1 #8 Myth: an "all-digital" signal path is best → It wastes amplifier power.

> Nothing digital exceeds 0 dBFS, yet cars need 6–12 dB more bass than the rest of the range, so never-clipped multi-kilowatt digital systems sound like 300 W ones. A simple analog chain with balanced outputs and differential inputs is fine. RARE
>
> **Source:** DIYMA 2017-07-05

In plain English: An "all-digital" path keeps the music as numbers from the source all the way to the amplifiers. Many people believe this is the best way. He says it wastes amplifier power. Nothing digital can go above 0 dBFS, the maximum digital level. Yet cars need 6 to 12 dB more bass than the rest of the range to sound natural. In a digital stream, he explains, you get that extra bass only by turning everything else down (or by giving the subs vastly more power than everything else), so in normal listening the amps deliver far less power than their ratings. The result, he says, is that multi-kilowatt digital systems that are never clipped end up sounding like 300 W systems. He says a simple analog chain is fine, as long as every part has balanced outputs and differential inputs (connections designed to cancel noise picked up along the cable).

CABLES, WIRING, POWER AND NOISE

5.1 #9 Myth: expensive or heavy cables improve sound → "You've been had".

> Directional and multi-shield cables are snake oil or band-aids for bad circuits; heavy-gauge RCAs do nothing because amp input impedance makes them unnecessary; of pure-silver power wire he said he needn't eat a foxglove salad to know it's poison. RARE
>
> **Sources:** DIYMA 2009-10-03 · DIYMA 2021-07-10 · DIYMA 2021-01-13

In plain English: Many people believe expensive or heavy cables improve sound. He says "You've been had". Directional cables (claimed to sound better one way round) and cables with many layers of shielding are, in his view, snake oil (fake cures) or band-aids for badly designed circuits. Heavy-gauge (thick) RCA signal cables do nothing, because the amp's input impedance makes them unnecessary. An amp's input draws almost no current, so a thin wire carries the signal just as well. Asked about pure-silver power wire, he said he needn't eat a foxglove salad to know it's poison. Foxglove is a poisonous plant; his point is that he does not need to try the product to know the claim is wrong.

5.1 #10 Myth: heavily shielded RCAs are always quietest → Match the cable to the input.

> Differential inputs want twisted pair, single-ended inputs want coax; test with an ohmmeter from RCA shield to amp ground (near 0 Ω = single-ended; hundreds of ohms or more = differential). Differential head-unit outputs reject noise just by twisting the wires. RARE
>
> **Sources:** CMA 2022 @50:01 · DIYMA 2021-11-26 · DIYMA 2021-09-25

In plain English: RCA cables carry the music signal from the radio or DSP to the amplifier. Many people believe the most heavily shielded RCA is always the quietest. He says to match the cable to the input instead. A differential input listens to the difference between two wires, so noise that lands on both wires equally cancels out; it wants twisted pair (two wires twisted together). A single-ended input measures the signal against ground; it wants coax (one wire inside a surrounding shield). His test: put an ohmmeter (a resistance meter) between the RCA's outer shield

and the amp's ground. Near 0 ohms means single-ended; hundreds of ohms or more means differential. If a head unit (the radio) has differential outputs, he says simply twisting the wires together rejects noise.

5.1 #11 Myth: speaker-level outputs are dirtier than RCA pre-outs → They are balanced and can be "far superior".

> They offer ground isolation and higher voltage; use them when a head unit lacks balanced RCAs (he finds it silly that aftermarket radios lack them).
>
> **Sources:** DIYMA 2013-01-15 · DIYMA 2021-09-24

In plain English: A factory radio often has only speaker-level outputs, the wires that normally drive the speakers. Aftermarket radios add RCA pre-outs, low-voltage outputs meant for amplifiers. Many people believe speaker-level outputs are the dirtier signal. He says they are balanced (neither signal wire is tied to ground) and can be "far superior". They offer ground isolation, which avoids noise caused by ground differences between devices, and a higher voltage, meaning a stronger signal. He says to use them when a head unit lacks balanced RCA outputs, and he finds it silly that aftermarket radios lack them.

5.1 #12 Myth: noise is fixed by re-routing, star grounds, 0-gauge cable or the "Big 3" → Differential inputs ended that era.

> Ground loops now usually come from mismatched input-preamp grounds, which is what a DSP's input-ground jumper is for; otherwise remove the offending device or add an isolator. Today PWM modules all over the car radiate noise, so cable type matters more than routing. RARE
>
> **Sources:** DIYMA 2008-12-21 · DIYMA 2021-09-24 · DIYMA 2021-11-03

In plain English: Many installers fight noise by re-routing cables, building a "star ground" (all grounds taken to one point), using huge 0-gauge cable, or doing the "Big 3" (upgrading the car's three main charging and ground wires). He says differential inputs ended that era. Today a ground loop (a noise caused by two grounds sitting at slightly different voltages) usually comes from a mismatch between one device's input ground and another's preamp ground. That is what the input-ground jumper on a DSP is for. Otherwise, he says, remove the device causing it or add an isolator. He adds that modern cars have PWM modules all over (electronics that switch power on and off very fast, for things like motors and lights), and these radiate noise, so the type of cable now matters more than how it is routed.

5.1 #13 Myth: hiss means a ground loop → Ground loops whine; hiss comes from gain structure, EQ boost or the amp.

> Amps with built-in DSP hiss more because their preamp scales every input down to an ADC limit of about 1–1.4 V; unplug the inputs to test. Auto-EQ boosting treble on a rolled-off factory source also makes hiss, so don't tear the install apart. RARE
>
> **Sources:** DIYMA 2015-09-16 · DIYMA 2021-10-14 · DIYMA 2021-08-08

In plain English: Many people hear hiss (a steady "shhh") and assume a ground loop. He says ground loops whine, while hiss comes from gain structure (how the level is set at each stage of the system), from EQ boost, or from the amp itself. Amps with a built-in DSP hiss more, because their preamp must scale every input down to the limit of their ADC (the chip that turns sound into

numbers), about 1 to 1.4 V. To test, unplug the amp's inputs: if hiss remains and falls when you turn the amp's gain down, the amp itself is the source. Auto-EQ that boosts the treble of a factory source whose treble is already rolled off (turned down) also makes hiss. So in practice: don't tear the install apart hunting for a ground problem.

5.1 #14 Myth: line drivers and high-voltage pre-outs improve sound → "Possibly the biggest waste of money ever invented".

> 30 V into an amp built for 4 V just clips its preamp; with a digital source start gains near 2 V. Exception: high voltage can beat radiated noise in very long runs.
>
> **Sources:** DIYMA 2010-11-26 · DIYMA 2015-06-20

In plain English: A line driver is a box that raises the signal voltage before it reaches the amplifier, and some radios advertise high-voltage pre-outs. Many people believe a higher voltage improves sound. He calls line drivers "possibly the biggest waste of money ever invented". Sending 30 V into an amp built for 4 V just clips (overloads) the amp's preamp. With a digital source, he says to start the gain settings near 2 V. His one exception: in very long cable runs, a high voltage can beat radiated noise (interference picked up from nearby electronics).

5.1 #15 Myth: cable, capacitors and charging upgrades must match the amp's fuse rating → Music draws a small fraction of that.

> Average music power is about 1/32 of peak (400 W peak ≈ 12.5 W average), mostly bass. A 2-farad cap isn't needed for 1,000 W (remove it if it blocks charging), and a 1-farad cap to limit sub bass costs as much as a DSP. RARE
>
> **Sources:** DIYMA 2021-12-22 · DIYMA 2021-08-02 · DIYMA 2021-08-19

In plain English: An amplifier's fuse is sized for its worst-case current draw. Many people believe the power cable, capacitors and charging upgrades must match that fuse rating. He says music draws only a small fraction of it. Average music power is about 1/32 of peak power, so an amp making 400 W peaks averages about 12.5 W, and most of that goes to the bass. A capacitor is a part that stores electrical charge and releases it quickly. He says a 2-farad capacitor isn't needed for a 1,000 W system, and if one is stopping you from charging the battery, remove it. And a 1-farad capacitor used as a filter to limit the bass going to a sub costs as much as a DSP.

5.1 #16 Myth: solder is the best connection → Crimp wire to terminal; solder only terminal to board.

> Spade lugs held old appliances together for 40 years; solder near the battery invites corrosion; heat on Audiofrog GS terminals can melt the plastic basket's internal joints. Tinned wire under a set screw cuts contact area, and "cinch terminals are not crimpers". RARE
>
> **Sources:** DIYMA 2021-07-21 · DIYMA 2021-11-26 · Audiofrog blog Dec 2020

In plain English: Many people believe soldering (joining metal with melted alloy) always makes the best connection. He says to crimp wire to a terminal (squeeze a metal connector onto it with the right tool), and to use solder only where a terminal joins a circuit board. Spade lugs, the push-on flat connectors, held old appliances together for 40 years. Solder near the battery invites corrosion. On Audiofrog GS speakers, heat on the terminals can melt the joints inside the plastic basket (the speaker's frame). Tinning wire (coating the strands with solder) before clamping it

under a set screw cuts the contact area. And "cinch terminals are not crimpers": with this he rejects the shortcut of letting a screw terminal squeeze a ferrule (a metal sleeve on the wire end) instead of crimping it properly.

5.1 #17 Myth: copper-clad aluminium is always inferior → Bad for power cable, good in voice coils

> (precise resistance; Audiofrog's flat CCA coils shed heat from both sides).
>
> **Sources:** DIYMA 2015-06-16 · CMA 2021 @19:00

In plain English: Copper-clad aluminium (CCA) is aluminium wire with a thin copper coating. Many people believe it is always worse than pure copper. He says it is bad for power cable but good in voice coils, the coil of wire that drives a speaker's cone. There it gives a precise resistance, and Audiofrog's flat CCA coils shed heat from both sides.

5.1 #18 Myth: "true 4-ohm" speakers matter, and amps "see" impedance → Nominal impedance is a label for choosing an amp; amps only know current.

> Audiofrog's GS "4 Ω" coils measure 2.3–2.8 Ω to suit factory and 2 Ω-rated amps; below rated impedance an amp is fine until its current limit, and modern amps then lower rail voltage rather than shut off. RARE
>
> **Sources:** DIYMA 2015-11-07 · CMA 2021 @33:05 · DIYMA 2022-02-14

In plain English: Impedance, measured in ohms, is roughly how much a speaker resists the current an amp sends it, and it changes with pitch. Many people believe speakers must be "true 4-ohm" and that amps "see" impedance. He says the nominal (labelled) impedance is only a label for choosing an amp; an amp only knows how much current it is delivering. Audiofrog's GS "4 Ω" speakers actually measure 2.3 to 2.8 Ω (ohms) so they suit factory radios and amps rated for 2 Ω. Below its rated impedance an amp is fine until it reaches its current limit, and modern amps then lower their rail voltage (internal supply voltage) rather than shut off.

AMPLIFIERS AND POWER

5.1 #19 Myth: high damping factor gives "superior cone control" → Anything above about 10 is enough.

> The voice coil's resistance sits in series with the amp's output impedance and dominates (DF 1000 moves Qts from 0.707 to 0.708); the story was invented to sell transistors over tubes, and makers quote it from habit. His view never changed, 2008–2021. RARE
>
> **Sources:** Audiofrog blog Dec 2020 · DIYMA 2008-11-28 · DIYMA 2021-05-08

In plain English: Damping factor (DF) is an amp spec: the speaker's impedance divided by the amp's own output impedance. Many people believe a high DF gives "superior cone control". He says anything above about 10 is enough. The speaker's voice coil has its own resistance, which sits in series with the amp's output impedance and dominates it. So even a DF of 1000 only moves Qts (a number describing how well a speaker's resonance is held in check) from 0.707 to 0.708. He says the story was invented to sell transistor amps over tube amps, and makers still quote DF out of habit. His view never changed from 2008 to 2021.

5.1 #20 Myth: slew rate is a separate quality spec → It already shows up in the power-bandwidth (THD) rating.

> An amp that can't swing full voltage at high frequencies shows treble distortion or must be rated only up to that frequency; swing at 30 kHz is irrelevant. RARE
>
> **Sources:** DIYMA 2017-07-07 · DIYMA 2017-07-05

In plain English: Slew rate is how fast an amp's output voltage can change. Many people treat it as a separate quality spec. He says it already shows up in the power-bandwidth rating, which states the power and THD (total harmonic distortion) across the audio range. If an amp can't swing its full voltage at high frequencies, that shows up as treble distortion, or the amp has to be rated only up to the frequency it can reach. Whether it can swing full voltage at 30 kHz, above human hearing, is irrelevant.

5.1 #21 Myth: class D for bass, class AB for highs → "Backwards"; there are good amps and bad amps.

> In randomized blind demos at CES 1996 almost nobody told a full-range class-D amp from a Krell, and the audible difference was in long bass notes. AB and D really differ only when clipped; judge a design, not the class (like condemning marinara after tasting one jar). RARE **Conflict:** three listeners after 700–800 demos (2022) vs four after about 1,500 (2023).
>
> **Sources:** CMA 2022 @29:01 · CMA 2023 @25:04 · DIYMA 2022-03-26

In plain English: Class AB and class D are two ways of building an amplifier; class D switches its output on and off very fast and wastes less energy as heat. Many people believe class D is for bass and class AB is for highs. He calls that "backwards": there are good amps and bad amps. At CES 1996 he ran randomized blind demos, where listeners didn't know which amp was playing, and almost nobody could tell a full-range class-D amp from a Krell (a high-end home amplifier). The audible difference was in long bass notes, which he credits to the Krell's huge storage capacitors, not to class D itself. He says AB and D really differ only when clipped (driven past their limit). So judge a design, not a class: condemning class D after one bad amp is like condemning marinara sauce after tasting one jar. The Conflict tag notes that his retellings differ: three listeners after 700 to 800 demos (2022) versus four after about 1,500 (2023).

5.1 #22 Myth: you can audition "amp sound", and pricier amps are more "dynamic" → You only hear an amp through a speaker.

> Unless the same difference appears on many speakers, it's a system difference (output impedance and class-D filters interact with the load). Judges called his car on 18 W chip amps "so dynamic"; he doesn't know what that means.
>
> **Sources:** CMA 2023 @19:05 · DIYMA 2013-01-05

In plain English: Many people believe you can audition an amp's own "sound", and that pricier amps are more "dynamic". He says you only ever hear an amp through a speaker. Unless the same difference shows up on many different speakers, what you heard is a difference in the whole system, because the amp's output impedance and a class-D amp's output filter interact with the speaker's load. Judges once called his car, running on 18 W chip amps (small amplifiers built on a single chip), "so dynamic", and he says he doesn't know what that means.

5.1 #23 Myth: amps "so good they need no feedback" → Like soles so thick you don't need shoes;

> feedback lowers noise, especially in the treble.
>
> **Source:** CMA 2022 @39:05

In plain English: Feedback in an amplifier means sending part of the output back to the input so the amp can correct its own errors. Some makers claim their amps are so good they need no feedback. He compares that claim to shoe soles so thick you don't need shoes. He says feedback lowers noise, especially in the treble.

5.1 #24 Myth: separate left/right amps (or better channel-separation specs) improve imaging → Car acoustics dwarf it.

> At 60 dB separation 1 W leaks a millionth of a watt, while each ear hears the opposite speaker at maybe −6 dB; "a car wash is more likely" to help. RARE
>
> **Sources:** DIYMA 2021-11-05 · DIYMA 2022-03-26

In plain English: Channel separation is how little of the left channel leaks into the right one inside the electronics, and the reverse. Imaging means hearing each voice and instrument coming from a clear spot. Many people believe separate amps for left and right, or better separation specs, improve imaging. He says the car's acoustics dwarf it. At 60 dB separation, 1 W in one channel leaks only a millionth of a watt into the other. Meanwhile, in a car each ear already hears the opposite side's speaker at maybe 6 dB quieter. In his words, "a car wash is more likely" to help.

5.1 #25 Myth: bridging causes phase shifts and each channel "sees half the load" → Neither.

> Bridging doubles voltage, so power rises 4×; "half the load" only means each side's current limit is unchanged. RARE
>
> **Sources:** DIYMA 2021-09-15 · DIYMA 2021-09-15b

In plain English: Bridging joins two amp channels to drive one speaker: one channel pushes while the other pulls. Many people believe this causes phase shifts and that each channel "sees half the load". He says neither is true. Bridging doubles the voltage across the speaker, and power rises with the square of voltage, so power goes up 4 times. He explains that "half the load" is just a way of saying each side's current limit is unchanged: each channel can still deliver only the same maximum current as before.

5.1 #26 Myth: too little power blows speakers → "You cannot blow a speaker with too little power."

> Clipping pushes a wave toward a square, whose average power is twice a sine's, so a clipped 100 W amp delivers about 200 W, all the extra at high frequencies. Class-D sub amps' output filters spare the sub, but a clipped radio or DSP sends that energy to the tweeters; he calls this the costliest misconception. RARE
>
> **Sources:** OFCAP 19 @20:07 · DIYMA 2021-07-01 · DIYMA 2021-07-08

In plain English: Many people believe an underpowered amp blows speakers. He says "You cannot blow a speaker with too little power." The danger is clipping: an amp pushed past its limit flattens the tops of the wave, turning it toward a square wave. A square wave carries twice the average power of a sine (smooth) wave of the same height, so a clipped 100 W amp delivers about 200 W,

and all the extra is at high frequencies. Class-D subwoofer amps have output filters that spare the sub. But a clipped radio or DSP passes that energy on to the tweeters. He calls this the costliest misconception.

5.1 #27 Myth: buy an amp with 2–3× the speaker's rating "for headroom" → Only safe if the gains can never clip it; "3× RMS" is nonsense.

> Setting treble gains with a −10 dB tone tells a 300 W amp to try for 3,000 W. The volume control, used with care ("carefully monitored", as pro audio says), is the real protection, and a warranty covers defects, not damage. RARE **Conflict:** in 2015 he called −10 dB/multitone gain setting (JBL's EZ gain) legitimate; the later warning targets maxing treble gains with such tracks.
>
> **Sources:** OFCAP 19 @27:00 · DIYMA 2021-09-20 · DIYMA 2021-07-10

In plain English: Many people buy an amp with 2 to 3 times the speaker's rating "for headroom" (spare power). He says that is safe only if the gains are set so the amp can never be clipped, and "3× RMS" is nonsense. His example: setting the treble gains with a test tone recorded at −10 dB tells a 300 W amp to try for 3,000 W, because full-level music is 10 dB louder than the tone, and 10 dB more means ten times the power. The real protection is the volume control, used with care ("carefully monitored", as pro audio says). And a warranty covers defects, not damage. The Conflict tag notes that in 2015 he called gain setting with −10 dB or multitone tracks (JBL's EZ gain) legitimate; his later warning is aimed at maxing out treble gains with such tracks.

5.1 #28 Myth: tweeters need 100–150 W channels → With music a tweeter gets a watt or two.

> Music's energy falls about 3 dB per octave like pink noise: calibrate an amp to 20 W and the band near 2.5 kHz gets about 0.16 W, so a 200 W amp at full output can cook a passive tweeter without clipping. His 24-channel car ran 30 W channels plus 800 W on subs. RARE
>
> **Sources:** Audiofrog blog Dec 2020 · DIYMA 2020-10-14 · OFCAP 19 @11:00

In plain English: Many people believe tweeters need 100 to 150 W amp channels. He says that with music a tweeter gets a watt or two. Pink noise is a standard test hiss whose level at any single frequency falls 3 dB (half the power) for each octave you go up, and he says music's energy falls in about the same way. So if you calibrate an amp to 20 W, the band near 2.5 kHz gets only about 0.16 W. The flip side: a 200 W amp at full output can cook a passive tweeter without ever clipping. In his worked example, the tweeter in a "20 W" passive speaker handles about 0.2 W, and a 200 W amp at full output feeds it about ten times that. His own 24-channel car ran 30 W channels, plus 800 W on the subs.

5.1 #29 Myth: passive crossovers waste lots of power, and active amplification is a band-aid for bad speakers → Both wrong.

> Passive loss is only the coils' series resistance (like asking what power an EQ cut "wastes": 6 dB is 75%); active crossovers just remove passive level-matching trade-offs. RARE
>
> **Sources:** DIYMA 2018-03-31 · DIYMA 2013-11-28

In plain English: A passive crossover is a set of coils and capacitors between the amp and the speakers that splits the frequencies. An active crossover splits them before the amps, usually in a DSP, with one amp channel per speaker. Many people believe passive crossovers waste lots of power, and that going active is a band-aid for bad speakers. He says both are wrong. The only real

loss in a passive crossover is the series resistance of its coils. He compares the question to asking how much power an EQ cut "wastes": a 6 dB cut means 75% less power at that frequency, even in a preamp EQ before the amp. Cutting is what a filter is for, and designers allow for the coils' resistance when they design the crossover. Active crossovers just remove the trade-offs a passive network makes to match the levels of the different drivers.

SPEAKERS AND DRIVERS

5.1 #30 Myth: speakers must "break in" → Only the spider loosens, and that matters mainly for subwoofers.

> Resin in the spider cracks, changing behaviour at and below resonance; mids, midbass and tweeters crossed above resonance barely change ("a nothingburger" for tweeters), and he re-measured a car three years on within 0.5 dB. Big changes signal a design flaw; liking a system more over time means you got used to it. RARE
>
> **Sources:** DIYMA 2017-07-28 · DIYMA 2021-05-07 · DIYMA 2015-05-30

In plain English: Many people believe new speakers must "break in", playing for hours before they sound right. He says only the spider loosens, and that matters mainly for subwoofers. The spider is the springy fabric ring behind the cone that keeps it centred. Resin in it cracks as it flexes, which changes the speaker's behaviour at and below its resonance (the note it naturally vibrates at). Mids, midbass and tweeters are crossed over above their resonance, so they barely change; for tweeters he calls it "a nothingburger". He re-measured a car three years on and it was within 0.5 dB. He says big changes signal a design flaw, and liking a system more over time means you got used to it.

5.1 #31 Myth: cone material gives a speaker its "sound" → Only stiffness, mass and damping matter, and an unsmoothed response graph shows whether they worked.

> "The right tone" is frequency-response management; renamed paper and exotic materials are marketing (to a seaweed-cone pitch he asked for MDMA cones for house music). He ranks it the second-biggest boondoggle after directional cables. RARE
>
> **Sources:** DIYMA 2015-12-23 · DIYMA 2021-09-14 · BusinessWorld 2024

In plain English: Many people believe the cone material, such as paper, metal or something exotic, gives a speaker its "sound". He says only stiffness, mass and damping (how well the cone soaks up its own vibrations) matter, and an unsmoothed frequency-response graph, one not averaged to hide the bumps, shows whether the designer got them right. To him, "the right tone" is frequency-response management. Renamed paper and exotic materials are marketing; when someone pitched him a seaweed cone, he joked by asking for MDMA cones for house music. He ranks cone material as the second-biggest boondoggle (a costly waste) after directional cables.

5.1 #32 Myth: a deeper cone has more effective area → It doesn't;

> displaced air depends on diameter (plus about a third to half the surround), not cone surface. RARE
>
> **Source:** DIYMA 2021-09-16

In plain English: Many people believe a deeper cone has more effective area, so it moves more air. He says it doesn't. The air a speaker displaces depends on its diameter, counting the cone plus about a third to half of the surround (the flexible rubber or foam ring at the edge), not on the

cone's surface area.

5.1 #33 Myth: phase plugs improve dispersion → They are "just plugs".

> Without a dust cap the cone's centre resonates like a bottle and the plug fills it; on a 6.5" it can lower a cone-mode peak but costs midband sensitivity.
>
> **Sources:** DIYMA 2015-10-09 · DIYMA 2021-12-04

In plain English: A phase plug is the fixed, bullet-shaped piece in the centre of some speakers, where a dust cap (the small dome over the centre) would normally sit. Many people believe phase plugs improve dispersion, meaning how widely the sound spreads. He says they are "just plugs". Without a dust cap, the open centre of the cone resonates like a bottle, and the plug fills that space. On a 6.5-inch speaker it can lower a peak caused by a cone mode (a frequency where the cone flexes instead of moving as one piece), but it costs midband sensitivity (loudness for a given power in the middle frequencies).

5.1 #34 Myth: billet baskets and "non-ringing" frames sound better → The basket contributes very little;

> big vents under the spider matter, and CNC machining is a low-volume cost and perceived-value choice.
>
> **Source:** DIYMA 2015-09-27

In plain English: The basket is the metal or plastic frame that holds a speaker's parts together. Some brands sell CNC-machined "billet" baskets, carved from a solid block of metal, or "non-ringing" frames as better-sounding. He says the basket contributes very little. What matters is big vents under the spider: air forced through small holes there makes mechanical noise. He says CNC machining is a choice driven by low production volume and by perceived value; in the same post he put the tooling for a cast 6-inch basket at about $5,000 to $20,000, a cost machining avoids.

5.1 #35 Myth: hand-built, "bespoke" or "individually matched" drivers beat "build house" products → Consistent production wins.

> Audiofrog tests 100% of units at end of line; "matched pairs" often signal a high-variance process; replicating a qualified design within tolerance is manufacturing wherever it happens, so "artisan" Europe vs "built not bought" USA is a double standard. RARE
>
> **Sources:** DIYMA 2013-11-28 · DIYMA 2015-05-31 · DIYMA 2021-06-29

In plain English: Many people believe hand-built, "bespoke" or "individually matched" drivers beat "build house" products (speakers from big factories that make drivers for many brands). He says consistent production wins. Audiofrog tests 100% of its units at the end of the production line. "Matched pairs" are often a sign of a high-variance process: you only need to hunt for matching pairs when units differ a lot. Copying a proven (qualified) design within tolerance, over and over, is manufacturing wherever it happens. So he mocks the double standard where a small European assembly line is called "artisan" and a US one "built not bought", while a factory elsewhere gets the "build house" label.

5.1 #36 Myth: "Made in Germany/USA" means quality and "made in China" means cheap labour → Quality is a specification, a test spec and verification.

> He has built products in about a dozen countries, each with great and poor factories, and labour is a small cost for line-built car audio. Audiofrog builds in China because only two factories would build to its reliability spec, and the chosen Tier-1 auto supplier cost more than any other. RARE
>
> **Sources:** DIYMA 2016-03-10 · DIYMA 2021-01-14 · DIYMA 2020-12-25

In plain English: Many people believe "Made in Germany" or "Made in USA" means quality, and "made in China" means cheap labour. He says quality is a specification, a test specification, and verification, meaning checking that products really meet them. He has built products in about a dozen countries, and each had both great and poor factories. Labour is a small part of the cost of car audio built on a production line. Audiofrog builds in China because only two factories would build to its reliability spec, and the one it chose, a Tier-1 auto supplier (a company that supplies parts directly to carmakers), cost more than any other.

5.1 #37 Myth: same factory means same driver, so buy "direct" on Alibaba → It depends what the brand contributed.

> Some brands only change the logo; others own tooling and specs, and small invisible changes shift performance a lot. Many brands hide country of origin (Dynaudio has been owned by China's GoerTek since 2014). RARE
>
> **Sources:** DIYMA 2022-03-29 · DIYMA 2022-03-31

In plain English: Many people believe that if two brands' drivers come out of the same factory they are the same driver, so you can skip the brand and buy "direct" on Alibaba. He says it depends on what the brand contributed. Some brands only change the logo. Others own the tooling (the moulds and dies that shape the parts) and the specifications, and small changes you can't see can shift performance a lot. He adds that many brands hide their country of origin; for example, Dynaudio has been owned by China's GoerTek since 2014.

5.1 #38 Myth: cone treatments such as EnABL help → In the posted data the untreated driver measured better every time.

> **Source:** DIYMA 2015-10-27

In plain English: EnABL is a treatment in which a pattern is painted onto a speaker's cone, with claims that it improves the sound. Many people believe treatments like this help. He says that in the test data that had been posted, the untreated driver measured better every time. In the same post he argued the pattern isn't random enough to change the cone's modes (the ways a cone flexes).

5.1 #39 Myth: tweeter-less "widebander" dash mids are purer → All cars sound better with tweeters.

> On axis widebanders give "clinical" pinpoint images with no space and off axis they sound dull; they work best in the dash aimed at the glass, as carmakers use them.
>
> **Sources:** DIYMA 2015-08-23 · DIYMA 2021-12-02

In plain English: A "widebander" (wideband or full-range driver) is a small speaker meant to cover almost the whole range of sound by itself, with no separate tweeter (the small speaker that plays the high notes). Many people believe a small widebander in the dashboard, with no tweeter, sounds "purer". He disagrees: all cars sound better with tweeters. Pointed straight at you ("on axis"), a widebander gives a "clinical" sound: each instrument seems to come from a tiny, exact spot, with no sense of space around it. Pointed away from you ("off axis"), it sounds dull, because the high notes fade off to the side. In his view widebanders work best the way carmakers use them: in the dash, aimed at the windshield glass.

5.1 #40 Myth: a point-source or coaxial driver is the ideal → A full-range point source is impossible, and a typical coaxial has no single "phase centre".

> A tweeter over a moving cone suffers intermodulation; a coax's real benefit is treble in the off-axis sound, and exact acoustic-centre alignment is "garbage" when heads move.
>
> **Sources:** OFCAP 19 @1:32:04 · DIYMA 2018-04-02

In plain English: A "point source" is a speaker whose whole sound comes from one single point. A "coaxial" speaker has a tweeter mounted in the middle of a bigger woofer, so both share one axis. Many people think that is the ideal design. He says a point source that plays the full range can't actually be built, and a typical coaxial doesn't have one single "phase centre" (one spot that all its sound seems to start from). Also, a tweeter sitting over a moving cone suffers "intermodulation": the cone's back-and-forth motion mixes with the treble and adds distortion. A coax's real benefit, he says, is that it puts treble into the off-axis sound, the sound that spreads sideways and bounces around the car. Lining up the exact acoustic centres is "garbage" to him, because real listeners move their heads.

5.1 #41 Myth: special tweeters (19 mm, ring radiators, "designed for off-axis") solve dispersion → Small gains where music has little energy.

> Same-diameter tweeters disperse alike; a smaller tweeter has a smaller coil in a market that gives speakers 3× their rating (he made 19 mm car tweeters at his previous company and expects big warranty costs). RARE
>
> **Sources:** DIYMA 2022-02-28 · DIYMA 2014-04-02

In plain English: "Dispersion" is how widely a speaker spreads its sound. Tweeters spread the highest notes less and less widely, so people look for special tweeters to fix that: very small 19 mm (about three-quarters of an inch) ones, "ring radiators" (a tweeter with a ring-shaped diaphragm), or ones marketed as "designed for off-axis". He says these give only small gains, and only in the very top notes, where music carries little energy. Tweeters of the same diameter spread sound alike. A smaller tweeter also has a smaller voice coil (the wire coil that drives it), so it handles less power. That matters in a market where people give speakers 3× their power rating. He made 19 mm car tweeters at his previous company, and he expects such tweeters would bring big warranty costs.

5.1 #42 Myth: high-end SQ speakers are delicate → "Not wedding cakes".

> Old high-end car speakers were untested home drivers; everyone plays loud on a Friday night, and

> the GB line was built for about 450 W.
>
> **Sources:** CMA 2023 @14:01 · OFCAP 19 @1:34:01

In plain English: "SQ" means sound quality. Many people believe expensive sound-quality speakers are fragile showpieces that must not be played loud. He says good high-end speakers are "not wedding cakes". He points out that older high-end car speakers were really home-audio drivers, never tested for life in a car. And everyone plays music loud sometimes, like on a Friday night. His company Audiofrog's high-end "GB" speaker line was built for about 450 W.

5.1 #43 Myth: a fat, higher-Q midbass is better bass → "That higher Q IS distortion."

> Door and kick locations are corners that already add about 6 dB, which is why the GB60 was voiced flat; a low-Q driver sounds thin on a board but full in the car.
>
> **Sources:** DIYMA 2011-08-27 · DIYMA 2017-08-17

In plain English: A "midbass" is the speaker, usually in the door, that plays the upper bass. "Q" is a number that describes how strongly a speaker resonates near its natural frequency. A higher-Q speaker has a hump of extra output there, which sounds "fat" and full. Many people take that fatness as better bass. He says that hump is the speaker adding something that isn't in the music, so it is distortion. In a car, the door and the kick panel (the panel by your feet) act like corners of a room, and corners already boost bass by about 6 dB (a clearly audible lift). That is why Audiofrog designed its GB60 midbass with a flat response. A low-Q speaker can sound thin on a store display board, yet sound full once it is in the car.

5.1 #44 Myth: dual-voice-coil wiring changes sensitivity or Q → It changes only the load impedance;

> Q depends on $(BL)^2/Re$ and both terms change together. RARE
>
> **Source:** DIYMA 2022-03-23

In plain English: Some subwoofers have two voice coils ("dual voice coil"). You can wire the two coils one after the other (series) or side by side (parallel). That changes the "impedance", the electrical load the amplifier sees, measured in ohms. Many people believe it also changes the speaker's sensitivity (how loud it plays from a given power) or its Q (how strongly it resonates). He says it changes only the impedance. Q depends on $(BL)^2/Re$. BL is the strength of the speaker's motor, and Re is the coil's resistance. When you rewire the coils, both of these change in step, so the result, and therefore Q, stays the same.

5.1 #45 Myth: a "subwoofer that plays midrange" exists → Efficiency, low Fs and small size don't go together

> (Hoffman's Iron Law); a light-coned low-Fs driver is fragile. RARE
>
> **Source:** DIYMA 2020-08-24

In plain English: Some people hope for one speaker that plays deep sub-bass and also clean midrange. He says that mostly doesn't exist. "Fs" is a speaker's resonance frequency: the low note at which it naturally vibrates most easily. A low Fs helps a speaker play deep bass. "Efficiency" is

how much sound you get per watt. "Hoffman's Iron Law" is a long-known rule of speaker design: you cannot have high efficiency, deep bass and a small size all at once. He adds that a speaker built with a light cone (good for midrange) and a low Fs (good for deep bass) ends up fragile.

5.1 #46 Myth: speakers are instruments with their own voice, and shorting rings are shields → Listeners prefer accurate speakers; shorting rings cut coil inductance.

> Pole vents exist mainly to cut noise behind the cone.
>
> **Sources:** DIYMA 2013-01-04 · DIYMA 2015-06-03

In plain English: This point covers three ideas. First, many people believe a speaker should be like a musical instrument, with its own pleasing "voice". He says listeners prefer accurate speakers, ones that play the recording without adding their own character. Second, a "shorting ring" is a metal ring (often copper) built into a speaker's magnet structure. Some believe it is a shield. He says its job is to cut the voice coil's "inductance", an electrical property of any coil that resists quick changes in current. Third, a "pole vent" is a hole through the centre of the magnet. He says it exists mainly to cut the noise made behind the cone.

5.1 #47 Myth: the box in the manual is a deliberately "safe", conservative design → Audiofrog's manual boxes are the best boxes.

> It designs woofers backwards from the car: in-car target first, then a convenient box size, then a driver that hits the target in that box. RARE
>
> **Source:** B1 Absolute Electronix 2023 @8:13

In plain English: A woofer's manual usually suggests a box (enclosure) to build for it. Many people believe these suggested boxes are deliberately cautious, "safe" designs rather than the best ones. He says Audiofrog's manual boxes are the best boxes for its woofers, because the company designs woofers backwards from the car. First it picks the sound it wants inside the car (the "in-car target"). Then it picks a box size that is convenient to fit. Then it builds a speaker that hits that target in that box. So for Audiofrog woofers, the box in the manual is not a compromise: it is the design.

INSTALLATION AND BASS (DETAIL IN SECTIONS 2 AND 4)

5.1 #48 Myth: foam rings "make midbass" and full door baffles help → Foam stops rattles;

> stick rings to the bracket, never the speaker, and trim baffles down to the rain shield.
>
> **Sources:** OFCAP 38 @34:02 · DIYMA 2021-09-21

In plain English: "Foam rings" are foam strips placed around a door speaker to seal it against the door panel, often sold as a way to get more midbass (upper bass). "Door baffles" are foam cups that fit behind the speaker. Many people believe both improve the bass. He says the foam's real job is to stop rattles. Stick the rings to the bracket (the mounting adapter that holds the speaker), never to the speaker itself. And cut the foam baffles down so only the top part remains as a rain shield, keeping water off the speaker, rather than using them fully sealed around it.

5.1 #49 Myth: deaden everything, and padding absorbs midbass → Deadening is rattle control.

> Absorbing ~200 Hz needs about 1.5 ft of material; a 250 Hz peak with no buzz is a reflection to EQ, not a panel to deaden.
>
> **Sources:** OFCAP 38 @1:13:01 · DIYMA 2021-09-07

In plain English: "Deadening" means applying sticky mats or similar material to a car's metal panels to stop them vibrating. Many people believe you should cover everything, and that padding soaks up excess midbass. He says deadening is for controlling rattles. Actually absorbing sound around 200 Hz would take about 1.5 ft (about 46 cm) of material. And if you measure a peak at 250 Hz but hear no buzzing or rattling, it is a reflection: sound bouncing inside the cabin and adding up at that one frequency. You fix that with EQ (an equalizer, which turns down just that frequency), not by deadening a panel.

5.1 #50 Myth: door midbass needs a sealed or ported door box → Not with a sub and an active high-pass,

> which does the box's job; venting a box into the door throws away the port's output. RARE
>
> **Sources:** DIYMA 2016-06-17 · DIYMA 2022-01-13

In plain English: Many people believe a door speaker needs its own box built inside the door, either sealed or ported (a "ported" box has an opening, called a port, that adds bass). He says that isn't needed when the system has a subwoofer and an "active high-pass": an electronic filter, set in the processor or amplifier, that stops the deepest bass from reaching the door speaker. That filter does the job the box would do. And if you build a ported box that vents into the door, the port's sound is lost inside the door, so you hear only the speaker. In a ported box you normally hear the speaker and the port together, so this throws away the port's whole benefit.

5.1 #51 Myth: vented boxes give "slow" bass → Group delay matters only above roughly 50 ms.

> "Sloppy" vented bass means a box tuned too high, and vented can lower distortion.
>
> **Sources:** DIYMA 2014-11-01 · DIYMA 2022-01-14

In plain English: A "vented" box is the same as a ported box: a speaker box with an opening that adds bass. Many people believe vented boxes give "slow" or late bass, which is bad for sound quality. "Group delay" is a measure of how much later some frequencies arrive than others. He says it only matters when it is really large, above roughly 50 ms (50 thousandths of a second). When vented bass sounds "sloppy", the box is tuned too high (its port is set for too high a note). He adds that a vented box can actually lower distortion. Elsewhere he explains that in a vented box the woofer itself moves less and distorts much less, though the port's own distortion can be higher.

5.1 #52 Myth: subs need a Qtc 0.707 box, small boxes are "tighter", and tuning below Fs blows subs → None holds in a car.

> The sub is low-passed below Fc, so box Q barely shows; a too-small box gives "one-note" bass; once boxed there is no Fs, only Fc or Fb, and cabin gain changes what you hear, not what the sub does.
>
> RARE
>
> **Sources:** DIYMA 2021-02-03 · DIYMA 2021-09-04 · DIYMA 2021-05-09

In plain English: This point knocks down three common beliefs about subwoofers. First, "Qtc" is the Q (resonance strength) of a sub in a sealed box, and 0.707 is often called the ideal value. He says that hardly matters in a car, because the sub is "low-passed" (filtered so it plays only the deep notes) below "Fc", the frequency where the boxed sub resonates. So the small effect of box Q barely shows. Second, many think a smaller box gives "tighter" bass. He says a box that is too small gives "one-note" bass instead. Third, many fear that tuning a vented box below the speaker's "Fs" (its own resonance frequency, outside a box) will blow the sub. He says that once a speaker is in a box there is no Fs any more, only Fc (sealed box) or Fb (the tuning frequency of a vented box). "Cabin gain", the extra deep bass a car's small cabin adds, changes what you hear, not what the sub itself is doing.

5.1 #53 Myth: a down-firing sub is "boundary loading" → It's an accidental bandpass box with unpredictable tuning.

> **Source:** DIYMA 2021-10-29

In plain English: A "down-firing" sub faces the floor. Many people say this uses "boundary loading", where a speaker is placed near a surface such as a floor or wall to gain bass. He says that's not what happens. The gap between the sub and the floor, plus the opening the sound escapes through, acts like a "bandpass" box: a box whose sound comes out only through an opening, which lets through just one band of frequencies. Because nobody designed that gap as a box, its tuning is accidental and you can't predict it.

5.1 #54 Myth: a midrange needs an enclosure, and the biggest driver that fits the pillar is best → Behind a high-pass the box barely matters, but a big driver in a small pod loses bass and peaks.

> He modelled the GB25 with a 200 Hz LR4 high-pass in boxes from about 0.16 L to infinite baffle and saw only slight differences; a pillar pod is an enclosure, so size the driver to it. RARE
>
> **Sources:** DIYMA 2017-07-02 · DIYMA 2017-10-01 · DIYMA 2021-11-19

In plain English: A "midrange" is a small speaker for the middle notes, often mounted in a "pod" on the A-pillar (the post beside the windshield). Many people believe it needs a sealed box behind it, and that you should fit the biggest speaker the pillar can hold. He says neither is right. With a "high-pass" filter in front of it (a filter that blocks the low notes), the box barely matters. He modelled Audiofrog's GB25 midrange behind a 200 Hz LR4 high-pass (a common, steep crossover filter type) in boxes from about 0.16 L (roughly 10 cubic inches) up to "infinite baffle" (no box at all). The results were only slightly different. But a pillar pod is itself a box. A big speaker crammed into a small pod loses bass and develops a peak, like a subwoofer in a box that's too small. So choose the speaker to suit the pod's size.

5.1 #55 Myth: cross door midbass at 40–50 Hz to pull bass forward → "50 Hz isn't midbass"; it's on his worst-mistakes list.

> High-pass the midbass about 80 Hz and low-pass the sub about 60 Hz (both LR4) so a 9–12 dB sub lift stays in phase; told mids are crossed at 40 Hz, he "stops listening". Bass "impact" lives near the mid-to-tweeter crossover. RARE
>
> **Sources:** OFCAP 19 @1:45:06 · DIYMA 2020-12-30 · DIYMA 2021-11-02

In plain English: The subwoofer usually sits in the back of the car, but listeners want the bass to seem to come from the front. A popular fix is to make the door speakers play very low, down to 40–50 Hz, so they carry more of the bass. He says "50 Hz isn't midbass", and he lists this among the worst mistakes. When someone tells him their door speakers are crossed at 40 Hz, he "stops listening". His recipe: filter the door midbass to play from about 80 Hz up (high-pass), and the sub to play from about 60 Hz down (low-pass), both with LR4 filters (a common steep crossover type). The sub is usually turned up 9–12 dB to sound natural, and this gap between the two filters keeps the sub and the door speakers "in phase" (pushing and pulling together) even with that boost. He also says bass "impact" (the punchy attack of a drum hit) lives near the crossover between the mid and the tweeter, which sits far above the bass notes themselves.

5.1 #56 Myth: with the windows down bass "waves can't fully develop" → Meaningless;

> the cabin simply turns from a leaky sealed box into a vented one. RARE
>
> **Source:** DIYMA 2018-03-23

In plain English: Many people explain why bass changes with the windows down by saying the long bass "waves can't fully develop" inside a small car. He calls that meaningless. With the windows up, the cabin acts like a leaky sealed box around the sub. Open a window and the cabin becomes a vented box, like a speaker box with a port. That is the real reason the bass changes.

CROSSOVERS, GAIN AND TUNING (DETAIL IN SECTION 3)

5.1 #57 Myth: steeper slopes protect tweeters better → LR4 protects best.

> Most power is removed in the first few dB below the corner, and a 48 dB/oct filter passes more near its knee; moving a tweeter from 4 to 2 kHz roughly triples its power. RARE **Conflict:** a round-4 note-taker could not square "LR4 beats 48 dB/oct" with simple attenuation one octave down; his knee argument is in the blog.
>
> **Sources:** Audiofrog blog Oct 2021 · DIYMA 2021-07-15

In plain English: A crossover filter keeps low notes away from a tweeter. Its "slope", in dB per octave (an octave is a doubling of frequency), says how fast it cuts the sound below its set point. Many people believe a steeper slope, such as 48 dB/oct, protects a tweeter better. He says LR4 (a common 24 dB/oct filter) protects best. His reason is the "knee" (or "corner"), the bend where the cut begins. Most of the power is removed in the first few dB of cut just below that corner, and right near its knee a 48 dB/oct filter lets more through than LR4 does. Separately, moving a tweeter's crossover from 4 kHz down to 2 kHz roughly triples the power it receives. The [Conflict] tag means this: one of the researchers compiling this collection could not square his claim with simple arithmetic, since one octave below the crossover a 48 dB/oct filter cuts more than a 24 dB/oct one. His own answer, the knee argument, is set out in his blog post.

5.1 #58 Myth: gentle 6 dB slopes "blend" better and phase shift makes crossovers bad → "Just more mythology".

> In cars LR4 beats both 6 dB/oct and steeper slopes; it sums flat and in phase, and its phase shift is intended.
>
> **Sources:** Audiofrog blog Jan 2021 · DIYMA 2021-09-27

In plain English: A 6 dB/oct slope is the gentlest crossover filter: two neighbouring speakers overlap over a wide range of notes. Many people believe this "blends" them better. Others believe crossovers are bad because they cause "phase shift", meaning the filter slightly shifts the timing of some frequencies. He calls both ideas "just more mythology". In cars, he says, LR4 (a 24 dB/oct filter) beats both gentle 6 dB/oct and steeper slopes. Its two halves add up to a flat, even response and stay in phase (in step) with each other. Its phase shift is intended: it is part of how the two halves fit together. His reasoning is that cars use high power and awkward speaker positions, so a narrow overlap blends better and gentle slopes protect too little.

5.1 #59 Myth: cross the tweeter (or any small driver) as low as it will go → Protect it first.

> Tweeters at about 1.5–2× resonance, everything except subs at or above Fs ("resonance IS distortion"); each octave lower costs about 2× the power and 4× the excursion.
>
> **Sources:** OFCAP 19 @1:41:04 · DIYMA 2022-03-10 · DIYMA 2017-08-17

In plain English: The "crossover point" is the note where a speaker starts playing. Many people set each small speaker as low as it can go, to get more out of it. He says a crossover's first job is to protect the smaller speaker. Every speaker has a "resonance" frequency (Fs), the note where it naturally vibrates most easily. He crosses tweeters at about 1.5–2× their resonance (for example, a tweeter that resonates at 1,000 Hz would start at 1,500–2,000 Hz). Every speaker except a subwoofer should be crossed at or above its Fs, because "resonance IS distortion". And each octave lower (each halving of the crossover frequency) costs about 2× the power and 4× the "excursion", meaning how far the cone has to move.

5.1 #60 Myth: active tweeters need no protection cap, or only audiophile caps → Add a cheap non-polar cap as a "parking brake".

> About 68 µF for a 4 Ω tweeter guards against amp failure and DSP typos; a resistor across it cures turn-on pops, and take it out if you return to a passive crossover. RARE
>
> **Sources:** Audiofrog blog Dec 2020 · DIYMA 2021-09-22

In plain English: In an "active" system, a DSP (digital signal processor, a small computer that splits and adjusts the sound) does the crossover work, and each speaker gets its own amplifier channel. A "capacitor" (cap) is a small electronic part that blocks low notes, so a cap wired to a tweeter protects it. Many people believe active tweeters need no cap, or that only expensive "audiophile" caps will do. He says to add a cheap "non-polar" cap (one that can be wired either way round) as a "parking brake": about 68 µF (microfarads, the unit of capacitance) for a 4 Ω tweeter. It protects the tweeter if an amplifier fails or if you make a typing mistake in the DSP settings. A resistor connected across the cap (from one leg to the other) cures the "pop" when the system turns on. If you later go back to a passive crossover (a box of parts that splits the sound with no DSP), take the cap out.

5.1 #61 Myth: set gains with a scope and 0 dB sines, and "max out" everything for resolution → Protect the DSP input first.

> Set it so it can never clip at full volume (4 V radio − set 5 V); digital clipping "sounds like broken speakers" and lands in mids and tweeters. Disconnect tweeters before playing 0 dB sines. RARE
>
> **Sources:** Audiofrog blog Mar 2021 · DIYMA 2021-08-04 · DIYMA 2020-10-14

In plain English: "Gains" are the input level settings on amplifiers and processors. A common method sets them with a "scope" (oscilloscope, a tool that draws the signal's wave shape) and "0 dB sines" (pure test tones at the maximum digital level). Many people also believe you should turn every level up as high as possible for better "resolution". He says protect the DSP's input first. Set it so it can never "clip" (run out of room and chop off the tops of the waves) even at full volume. For example, if the radio puts out at most 4 V (volts), set the DSP input for 5 V. Digital clipping "sounds like broken speakers", and the harsh distortion lands in the mids and tweeters. And disconnect the tweeters before you play 0 dB test tones, because full-level tones can destroy them.

5.1 #62 Myth: delay widens the stage or moves the centre → Delay only plays a speaker later; move the image with level.

> Set delays by distance, then use balance: 6 dB moves the centre halfway to one side. Delay "tweaks" just add comb filtering, and hard-panned sound comes from its speaker (pan fully left and point). RARE
>
> **Sources:** Audiofrog blog Dec 2020 · DIYMA 2017-08-18 · DIYMA 2018-05-22

In plain English: "Delay" (time alignment) is a DSP setting that holds back the sound from the speakers nearest you, so the sound from every speaker reaches your ears together. The "stage" is the illusion of musicians spread across the dashboard, and the "centre" is where a singer seems to stand. Many people believe tweaking delays makes the stage wider or moves the centre. He says delay only makes a speaker play later. His blog's analogy: someone leaving home later doesn't move their house. Set delays from measured distances. Then move the image with level, using the balance control: a 6 dB level difference moves the centre halfway to one side. Delay "tweaks" only add "comb filtering": when the same sound arrives from two places at slightly different times, some notes cancel and others add up, leaving a row of dips like the teeth of a comb. A sound recorded in only one channel ("hard-panned") comes from that channel's speaker. To find your stage's edge, pan the balance fully left and point to where the sound is.

5.1 #63 Myth: delays need millimetre precision from impulse peaks, lasers or cross-correlation → A tape measure is as good.

> An impulse peak is not the arrival time (for subs and dash speakers it is plainly wrong); a one-sample error at 48 kHz is out of phase only at 24 kHz, and a $100 cross-correlation method lands no closer. He was accused of taking SQ "back 20 years" for this. RARE
>
> **Sources:** D&F live @15:07 · DIYMA 2021-03-26 · DIYMA 2021-09-02

In plain English: Many people believe delays must be set with great precision, using measuring software (reading the peak of an "impulse response", a recorded test click), laser distance meters or "cross-correlation" (a maths method that compares two signals to find the time gap). He says a tape measure is just as good. The peak of an impulse measurement is not the moment the sound arrives; for subwoofers and dash speakers it is plainly wrong. A DSP running at 48 kHz works in steps of one "sample" (1/48,000 of a second, about 7 mm of sound travel). Being off by one sample only puts two speakers fully out of phase (cancelling) at 24 kHz, above human hearing. And a $100 cross-correlation method gets no closer than a tape measure. Some accused him of taking sound quality "back 20 years" for saying this.

5.1 #64 Myth: delay should also correct crossover lag, reflections or cabin gain → Delay corrects distance only.

> As long as there is a direct path, the direct sound arrives first and sets the delay. RARE
>
> **Sources:** DIYMA 2021-03-26 · DIYMA 2021-03-26b

In plain English: Many people try to use delay to fix other things too: the small timing lag a crossover filter adds, sound bouncing off the glass and panels (reflections), or "cabin gain" (the extra deep bass a small car cabin adds). He says delay corrects distance only. As long as there is a straight, unblocked line from the speaker to your ear, the direct sound arrives first. Reflections always come later, so the direct sound is what sets the delay.

5.1 #65 Myth: half a wavelength of delay plus a polarity flip brings bass forward → It fixes phase at one frequency only.

> Bass won't come from the front when high-pass and low-pass phase aren't complementary; keep the acoustic crossover at 80 Hz by lowering the LR4 sub low-pass half an octave per 12 dB of sub boost. RARE
>
> **Sources:** DIYMA 2021-08-19 · DIYMA 2021-08-21

In plain English: A trick some tuners use to make bass seem to come from the front: delay the sub by half a wavelength (half of one cycle of a bass note) and flip its "polarity" (swap its plus and minus wires so the cone moves the other way). He says that only fixes the phase at one single frequency. Bass won't sound like it comes from the front when the "phase" (timing within each cycle) of the front speakers' high-pass filter and the sub's low-pass filter aren't "complementary", meaning made to fit together. His recipe: keep the "acoustic crossover", where the sub and front speakers actually meet as measured in the car, at 80 Hz. To do that, lower the sub's LR4 low-pass by half an octave for every 12 dB you turn the sub up. (Half an octave below 80 Hz is about 57 Hz.) He adds that this works if the slopes were EQ'd close to their target shapes during tuning.

5.1 #66 Myth: flip polarity to fix imaging → A pre-DSP band-aid

> that moves the door-speaker cancellation from 250 to 500 Hz; never combine it with delay.
>
> **Source:** Audiofrog blog Dec 2020

In plain English: "Imaging" is how well you can place each sound, like the singer in the centre. The driver sits much closer to one door speaker than the other, so their sounds reach the ear at different times, and at about 250 Hz they cancel each other. Many people "fix" imaging by flipping the polarity of one side (swapping its plus and minus wires). He calls this a band-aid from before DSPs were common. It doesn't remove the cancellation: it only moves it from 250 Hz to 500 Hz. And never combine a polarity flip with delay. In the same blog post he says about 2 ms (two thousandths of a second) of delay on the near side fixes it for one seat.

5.1 #67 Myth: phase coherence (FIR filters, phase at the tweeter crossover) is the holy grail → "The new snake oil".

> Phase at the mid-tweeter crossover can't be measured in a way that matches hearing, and FIR is not inherently "frequency-dependent delay"; what matters is sub-midbass and midbass-mid phase far

> from 180°. (He does use targeted all-pass filters for two seats.)
>
> **Sources:** DIYMA 2013-12-26 · DIYMA 2021-07-12 · DIYMA 2021-08-20

In plain English: "Phase" describes where each sound wave is in its up-and-down cycle when it reaches you. "Phase coherence" is the idea that every frequency should arrive perfectly in step. Many people treat it as the ultimate goal, chasing it with "FIR" filters (a type of digital filter) or by perfecting the phase where the mid hands over to the tweeter. He calls it "the new snake oil" (a product sold on false promises). At the mid-to-tweeter crossover, the waves are so short that phase can't be measured in a way that matches what you hear. And FIR filters are not, by their nature, "frequency-dependent delay"; he says they can adjust phase separately, but that isn't built into the filter type. What matters is the phase between sub and midbass, and between midbass and midrange: it should be far from 180° (exactly opposite, which cancels). He does use targeted "all-pass" filters, which shift phase without changing loudness, to make a tune work in two seats.

5.1 #68 Myth: aim (or laser-aim) mids at the listener → Drivers are omnidirectional below beaming;

> only tweeters above about 10 kHz respond to aiming.
>
> **Source:** Audiofrog blog Jun 2021

In plain English: Many people carefully aim their midrange speakers at the listener, sometimes even with a laser. He says aiming them achieves nothing. A speaker sends low and middle notes out in all directions ("omnidirectional"). Only above a certain frequency, which depends on its size, does its sound narrow into a beam ("beaming"). Below that point, aiming has no effect. In a car, he says, only tweeters above about 10 kHz respond to aiming.

5.1 #69 Myth: one target curve is much better than another → Good car curves differ by 1–2 dB;

> big claimed differences mean neither was reached.
>
> **Source:** Audiofrog blog Mar 2021

In plain English: A "target curve" is the planned balance of bass, midrange and treble that a tuner aims for when adjusting a system. Many people argue that one curve is far better than another. He says good car target curves differ by only 1–2 dB, which is small. If someone claims a big difference between two curves, it means neither curve was actually reached in the car.

5.1 #70 Myth: you need an expensive mic, the right 0°/90° file and seated measurements → Calibrated mics mostly agree.

> Eleven of his twelve mics agree within about 1 dB (avoid the APEX 220); a car is a diffuse field where mic orientation hardly matters; measure from outside, because your body skews the treble. RARE
>
> **Sources:** UMI-1 video @9:01 · DIYMA 2021-08-16 · DIYMA 2020-10-03

In plain English: Tuners measure a car's sound with a measurement microphone. A "calibrated" mic comes with a correction file, often in two versions: "0°" (mic pointed at the speaker) and "90°" (mic pointed up). Many people believe you need an expensive mic, the right file, and measurements taken while you sit in the seat. He says calibrated mics mostly agree: eleven of his twelve measure within about 1 dB of each other, and he says to avoid the APEX 220. A car is a "diffuse field": with so

many reflections, sound reaches the mic from all directions at once, so the way the mic points hardly matters. And measure from outside the car, because your body inside it skews the treble readings.

5.1 #71 Myth: more precision is better (hundreds of mic positions, 1/24–1/48-octave EQ, delays in hundredths of an inch) → Imported pro/home mythology.

> Five or six positions around the head are enough; don't EQ the treble comb filter, which shifts with an inch of head movement. RARE
>
> **Sources:** DIYMA 2021-08-06 · DIYMA 2020-07-08 · DIYMA 2021-08-26

In plain English: Many tuners believe more precision always helps: measuring at hundreds of mic positions, EQ adjustments only 1/24 or 1/48 of an octave wide (very narrow slices of the sound), and delays set to hundredths of an inch. He calls this mythology imported from professional sound and home audio, methods that don't fit a car. Five or six mic positions around the head are enough. And don't use EQ on the "comb filter" in the treble, the dense pattern of narrow peaks and dips caused by sound arriving from several places at slightly different times. It shifts if your head moves just one inch (about 2.5 cm), and he adds that we don't really hear it.

5.1 #72 Myth: tune with 31 sine tones and centre the image with pure tones → "A really bad idea".

> Above 1 kHz a tone easily lands in a comb-filter null that differs left and right; use band-limited pink noise or music. RARE
>
> **Sources:** DIYMA 2021-07-08 · DIYMA 2020-07-08

In plain English: An old competition habit is to tune with 31 "sine tones" (pure single-pitch test tones, one per band of a 31-band equalizer) and to centre the image using pure tones. He calls this "a really bad idea". Above 1 kHz, a single tone can easily land in a "null" (a deep dip where sound cancels) of the car's comb filter. That null sits in a different place for the left and the right speaker, so the tone misleads you. Instead, use band-limited pink noise (a hiss-like test sound with equal energy in every octave, filtered to cover just the range you are checking) or music.

5.1 #73 Myth: uncorrelated pink noise is the "better" test signal → It has no centre, so it can't check imaging.

> Use mono (correlated) noise; uncorrelated noise is only for gaming an IASCA RTA lane or muting the centre in an upmixer. RARE
>
> **Sources:** DIYMA 2020-07-08 · DIYMA 2021-05-09 · DIYMA 2022-01-31

In plain English: Pink noise test signals come in two kinds. "Correlated" (mono) noise is identical in the left and right channels. "Uncorrelated" noise is different random noise in each channel. Many people think uncorrelated noise is the better test signal. He says it has nothing common to both channels, so it has no centre image, and so it can't check imaging (whether sounds appear in the right places). Use mono noise. He sees only two uses for uncorrelated noise. One is "gaming" the RTA lane of an IASCA contest (IASCA runs car-audio competitions, and an RTA, or real-time analyzer, measures the system's response there). The other is making every speaker except the centre play when tuning an "upmixer", a processor that creates centre and rear channels from stereo.

5.1 #74 Myth: a good passenger seat must cost the driver's seat → Only when delay is your only tool.

> All-pass filters, an upmixer with a real centre, or kick-panel placement (seat-to-seat path difference under about a foot) serve both seats. RARE
>
> **Sources:** SQOLOGY Pt 2 @6:02 · DIYMA 2021-06-29

In plain English: Delay can make the sound from all speakers arrive together at one seat only, usually the driver's. So many people believe that making the passenger seat sound good must make the driver's seat worse. He says that is only true when delay is your only tool. Other tools serve both seats: "all-pass" filters (which shift phase without changing loudness), an upmixer that creates a real centre channel, or placing speakers in the kick panels (the panels by the occupants' feet). Kick panels work because they keep the difference in distance between the two seats under about a foot (about 30 cm).

5.1 #75 Myth: an L+R "centre" speaker or L−R rear fill works → An L+R centre is worse than none, and he now steers people away from L−R rear fill too.

> The real centre is only the content common to both channels; for rear fill use plain stereo, high-passed at 100 Hz or more and delayed about 12–20 ms. **Changed view:** in 2017 he gave DIY L−R rear-fill recipes; by 2021 he calls difference rear fill "phasy" and says he may have helped make it semi-mythical.
>
> **Sources:** DIYMA 2015-08-05 · DIYMA 2021-07-25 · DIYMA 2021-12-07

In plain English: Two DIY tricks are common. One feeds a centre speaker in the dash with "L+R", the left and right channels mixed together. The other feeds the rear speakers ("rear fill", quiet rear sound for a sense of space) with "L−R", only what differs between left and right. He says an L+R centre is worse than having no centre at all. The real centre of a recording is only the content common to both channels, while a plain L+R mix also includes sounds that are only on one side. For rear fill, he now says to use plain stereo, filtered so the rears play only from 100 Hz or more upward and delayed about 12–20 ms (thousandths of a second). The [Changed view] tag means his advice changed over time. In 2017 he gave do-it-yourself recipes for L−R rear fill. By 2021 he called difference rear fill "phasy" (strange-sounding to him) and said he may have helped give it its semi-mythical status on the forum.

5.1 #76 Myth: "cars do strange things" and "every car is different, just play with it" → Cars are predictable physics.

> When tape-measured delays don't give the expected phase, something else is affecting phase; the tape measure isn't broken. Skipping basics turns tuning into a Rube Goldberg machine. RARE
>
> **Sources:** DIYMA 2021-08-21 · DIYMA 2021-12-22

In plain English: Many people explain odd results by saying "cars do strange things" or "every car is different, just play with it until it sounds better." He says cars follow predictable physics. If delays set with a tape measure don't give the phase you expected, the right conclusion is that something else is affecting the phase. The tape measure isn't broken. In his view, skipping the basics turns tuning into a "Rube Goldberg machine": a needlessly complicated contraption that does a simple job the long way round. His examples of this are delaying half a wavelength and then flipping polarity, or "playing with the delays" to widen the stage.

5.2 Reading specs, measurements, reviews and claims

What this part is about: Reading the numbers, graphs, reviews and sales claims you meet when shopping for car audio: power ratings, amplifier specs, speaker data sheets and measurements. His general message is to ask what a number really measures, how it was tested, and what the physical mechanism behind a claim is.

5.2 #1 A speaker's power rating means nothing without its test filter.

> Ratings use pink noise with a high-pass for tweeters and mids, so a "100 W tweeter" handles 100 W of high-passed signal (Audiofrog's GB10 sheet pairs 100 W with a 2.5 kHz, 24 dB/oct high-pass).
>
> **Sources:** Audiofrog blog Dec 2020 · GB10OE sheet (COMPANY)

In plain English: A watt rating on a speaker describes a test, not a promise that the speaker can take that much of any signal. Pink noise is a hiss-like test signal containing all frequencies (all notes), with more energy in the low notes. A high-pass filter blocks low notes and lets high notes through. Tweeters (small speakers for high notes) and mids are normally rated with such a filter in front of them, so much of the pink noise's energy never reaches them. His point is that a "100 W tweeter" handles 100 W of that filtered signal. Audiofrog's GB10 tweeter sheet, for example, gives 100 W together with a 2.5 kHz (2,500 cycles per second), 24 dB per octave high-pass, a steep filter. So the watt number only makes sense alongside the filter it was tested with.

5.2 #2 Read "RMS" as continuous average power plus a separate peak rating, and treat both as limits, not requirements.

> Speakers die from heat or mechanical peaks and have no minimum power; Audiofrog's 100-hour test uses shaped noise matching both ratings, and mechanical limits need their own test (Don Keele's ramped-sine "boink" test). RARE
>
> **Sources:** OFCAP 19 @26:02 · OFCAP 19 @39:01 · DIYMA 2021-09-19

In plain English: "RMS" is the power figure most people look at on a speaker. He says to read it as the continuous average power the speaker can take, and to look for a separate peak figure for short bursts. Both are ceilings, not amounts you must supply: a speaker has no minimum power. Many people believe a speaker "needs" its rated watts to work properly; he disagrees. A speaker dies in one of two ways. It overheats (its voice coil, the wire coil that drives the cone, gets too hot), or a big peak pushes it past its mechanical limits. Audiofrog's own test runs for 100 hours with a noise signal shaped so that its average matches the RMS rating and its peaks match the peak rating. Mechanical limits need a test of their own. He names Don Keele's "boink" test, which uses a ramped sine (a single pure tone that steadily gets bigger) to test the mechanics without heating the coil.

5.2 #3 Sanity-check amp power claims against the fuse and the price.

> A 20 A fuse on a "2000 W" amp, or 2000 W for $99, can't be true; car amps are roughly constant-voltage, so "doubling into 2 Ω" is theory, not a given.
>
> **Sources:** DIYMA 2011-12-16 · DIYMA 2013-04-23

In plain English: A fuse is a safety part that blows if an amp draws too much current from the car's electrical system. An amp can't put out more power than it takes in, so a small fuse and a huge watt claim don't fit together. His examples: a 20 A fuse on a "2000 W" amp, or 2000 W for $99, can't be true. He also explains a common spec trick. Ohms (Ω) measure a speaker's electrical load; lower ohms lets more current flow. Car amps behave roughly like a fixed-voltage source, and in theory the same voltage into half the load (2 Ω instead of 4 Ω) gives double the power. He says that doubling is only theory, not something an amp is guaranteed to deliver. So in practice: check the claimed watts against the fuse size and the price.

5.2 #4 A heavy heatsink signals sustained power; small amps often roll back.

> Some amps cut output within minutes at rated power, so "150 W" behaves like 75 W; many tiny amps now cut rail voltage when hot instead of using aluminium. RARE
>
> **Sources:** CMA 2022 @1:09:04 · DIYMA 2022-01-14

In plain English: Amplifiers turn part of their power into heat. A heatsink is the finned metal body, usually aluminium, that carries the heat away. His point is that a big, heavy heatsink is a sign the amp can keep up its power for a long time. Many small amps can't. They "roll back", meaning they cut their output when they get hot. He says some do it within minutes at their rated power, so an amp sold as "150 W" behaves like a 75 W one. "Rail voltage" is the internal supply voltage an amp uses to make its output; lowering it lowers the power available. Many tiny amps now protect themselves this way when hot, instead of carrying lots of aluminium. Many people take a small amp's watt rating at face value; he says to look at how it deals with heat.

5.2 #5 Compare amps' distortion and noise at about 1 W, not at rated power.

> A tweeter lives near a watt, some amps distort more at very low power, and hiss matters more in a car because you sit close to tweeters. RARE
>
> **Sources:** DIYMA 2021-05-18 · CMA 2023 @22:05

In plain English: Amp spec sheets usually quote distortion (unwanted changes the amp adds to the signal) and noise at full rated power. He says to compare amps at about 1 watt instead, because that is close to what a tweeter actually receives with music. Some amps distort more at very low power than at high power, and a full-power figure hides that. Noise matters more in a car too. Hiss is the steady "shhh" that comes from the speakers even with no music playing, and in a car you sit very close to the tweeters, so you hear it. Many people compare amps by their full-power numbers; his point is that the figures near 1 W tell you more about how the amp will sound.

5.2 #6 Class-D treble response depends on the speaker load.

> A class-D output filter is designed for one load like a passive crossover, so 2 Ω vs 4 Ω gives a roll-off or a peak; Audiofrog moved its filter to 40 kHz. Real, he says, but not what separates great cars from bad ones. RARE
>
> **Sources:** CMA 2022 @35:07 · B1 @29:58

In plain English: Class-D is the efficient amplifier design used in most small car amps. It works by switching on and off very fast, and it needs a filter on its output to remove that switching signal. A passive crossover is the set of coils and capacitors that splits sound between speakers, and it only works as designed with the speaker it was designed for. He says a class-D output filter is the same:

it is designed for one load. Connect a different load, such as 2 Ω instead of 4 Ω, and the top of the treble either falls away (a roll-off) or rises (a peak). Audiofrog moved its filter up to 40 kHz (40,000 cycles per second, well above the roughly 20 kHz upper limit of human hearing) so the effect lands outside what you can hear. He says the effect is real, but it is not what separates great-sounding cars from bad ones.

5.2 #7 Small class-D amps: ask about FM interference, hiss and layout.

> Tiny amps hiss or whine because the supply transformer's field sits near the preamp; look for shielding, four-layer boards and careful transformer placement. Nothing on the box warns you. RARE
>
> **Sources:** CMA 2023 @26:04 · B1 @26:01

In plain English: Very small amps pack everything tightly together. He says that is why many hiss or whine: the magnetic field from the power-supply transformer (the part that converts the car's 12 volts into what the amp needs) sits close to the preamp, the sensitive first stage that handles the weak incoming signal. Class-D switching can also interfere with FM radio reception. What helps is shielding (metal that blocks the interference), four-layer circuit boards (boards with extra internal layers, which can include a ground layer acting as a shield) and careful placement of the transformer. Nothing on the box warns you about any of this. So in practice: before buying a small class-D amp, ask about FM interference, hiss and how it is laid out.

5.2 #8 Amps with built-in DSP are convenience products with a gain-structure trap.

> Big inputs are attenuated to the converter's limit, and the output DAC can't exceed 0 dBFS with no analog gain after it, so after tuning cuts a "1,000 W" amp may deliver about 10 W. For high-end systems use a separate DSP and amps with analog input controls. RARE
>
> **Sources:** B1 @33:35 · B1 @36:46

In plain English: A DSP (digital signal processor) is the small computer that handles crossovers, delays and EQ (tone adjustments) in a modern system. Some amps have one built in. He calls these convenience products and warns of a "gain-structure" trap. Gain structure means how the signal level is set at each stage of the chain. The trap, step by step. First, the input: the converter that turns the incoming signal into numbers can only accept a limited voltage, so a big input signal has to be turned down to fit. Second, the output: the DAC (digital-to-analog converter) that turns the numbers back into a signal cannot go above 0 dBFS, the digital maximum ("full scale"). In these amps there is usually no analog volume control after that converter, so nothing can raise the level later. Third, tuning: tuning mostly cuts levels. With nowhere to get the level back, he says, a "1,000 W" amp may end up delivering about 10 W. So in practice: for high-end systems he recommends a separate DSP plus amps with analog input (gain) controls.

5.2 #9 Know the sensitivity arithmetic.

> Compare at 2.83 V (car amps are voltage sources); 80–125 W is only +1.9 dB, and about 99% of amp power becomes heat. A sub's sensitivity is a midband number that doesn't predict in-car level, and GB tweeters are 6 dB hotter on purpose so you can turn gains down and cut hiss. RARE
>
> **Sources:** DIYMA 2015-11-26 · DIYMA 2022-01-26 · DIYMA 2017-10-01

In plain English: Sensitivity is how loud a speaker plays for a given input. A dB (decibel) is the unit for differences in loudness. He says to compare speakers at 2.83 volts rather than "1 watt", because car amps act as voltage sources: they deliver a set voltage, and the watts that flow depend on the speaker's ohms. (2.83 V gives 1 W into 8 Ω or 2 W into 4 Ω.) Some arithmetic he gives: going from 80 W to 125 W adds only 1.9 dB, a small step in loudness. And a speaker turns only about 1% of the amp's power into sound; about 99% becomes heat. A subwoofer's sensitivity figure is measured in its middle range and doesn't predict how loud it will be in a car. Finally, Audiofrog's GB tweeters are made 6 dB more sensitive on purpose, so you can turn the amp's gain down, which also cuts hiss. Many people chase more watts; his numbers show how little extra watts buy.

5.2 #10 Efficiency vs power: context, not reversal.

> In 2008–2013 "efficiency is overrated, power is cheap" argued for active crossovers; in 2021 he stressed efficiency for very loud systems, where heavy coils lower efficiency further.
>
> **Sources:** DIYMA 2013-11-28 · OFCAP 19 @33:07

In plain English: Efficiency is how much sound a speaker makes from a given amount of power. Two statements of his can look like opposites, and this point explains that they answer different questions. Between 2008 and 2013 he said "efficiency is overrated, power is cheap". That was an argument for active crossovers: giving each speaker its own amp channel and setting levels electronically, instead of juggling resistors and parts in a passive crossover to balance the speakers' levels. In 2021 he stressed efficiency for very loud systems. There, the heavy voice coils needed to take huge power make the speakers less efficient still, so efficiency matters more.

5.2 #11 Thiele/Small parameters tell box fit and low-frequency roll-off, not quality.

> "Low-Q woofers are more accurate" is mostly wrong; BL alone means nothing; and Qts = Qes·Qms/(Qes+Qms), not a product.
>
> **Sources:** DIYMA 2009-10-23 · DIYMA 2021-05-09

In plain English: Thiele/Small (T/S) parameters are a standard set of numbers describing how a woofer behaves at low frequencies, such as its resonance and its "Q" values. He says they tell you what box suits the driver and how its bass rolls off (fades out) at the bottom. They don't tell you whether it is a good speaker. Q describes how strongly the speaker's resonance is held in check. Many people believe "low-Q woofers are more accurate"; he says that is mostly wrong. BL, the strength of the motor (magnet and coil), means nothing on its own. He also corrects a formula people get wrong: total Q (Qts) is Qes × Qms divided by (Qes + Qms), the same maths as two resistors in parallel, not simply Qes × Qms.

5.2 #12 Complete speaker data is enough to judge a driver better than ears.

> T/S parameters, on-axis plus 15/30/45/60° responses and 2nd/3rd-harmonic distortion; check that off-axis angles match before comparing sheets. Klippel data mainly saves designers time, and specs reveal a driver's intended job. RARE
>
> **Sources:** DIYMA 2016-01-29 · DIYMA 2021-12-24 · DIYMA 2022-01-24

In plain English: A driver is a single speaker unit (a woofer, mid or tweeter). He says a complete set of data lets you judge a driver better than your ears can. Complete means: the T/S parameters; the frequency response (a graph of how loud it plays each note) straight ahead, called on-axis, plus at

15, 30, 45 and 60 degrees to the side; and its second- and third-harmonic distortion (unwanted extra tones at two and three times the note being played). Before comparing two data sheets, check that both measured at the same off-axis angles. Klippel is a specialised speaker-measurement system; he says its data mainly saves designers time. Specs also reveal what job a driver was made for. Many people believe only listening can tell you whether a speaker is good; he disagrees.

5.2 #13 Red flags on a spec sheet.

> Missing curves, 1/3-octave smoothing, impossible wiggles where cones are still pistonic (near 80 Hz on a 6.5"), sub curves flat to the top; about 10% tolerance on suspension parameters is normal.
>
> **Sources:** DIYMA 2015-09-30 · DIYMA 2015-05-31

In plain English: A spec sheet is the maker's data page for a speaker. He lists warning signs. Graphs (curves) that are missing. Curves shown with 1/3-octave smoothing, which blurs away detail the sheet should show. Wiggles (peaks and dips) where they are physically impossible: a cone is "pistonic" when it moves all in one piece, like a piston, and in that range its response is smooth, so wiggles near 80 Hz on a 6.5-inch speaker are a red flag. And subwoofer curves shown flat all the way to the top of the graph; the sources list this as a red flag without spelling out the reasoning. On the other hand, about 10% variation from unit to unit in suspension parameters (those set by the springy parts that hold the cone) is normal, not a defect.

5.2 #14 Read distortion by type and shape, not just level.

> Even-order comes from asymmetry (often the spider) and masks detail; odd-order comes from the coil leaving the gap and sounds mechanical. Convert with 20·log; near-field measurements miss cone breakup, so measure far enough away. RARE
>
> **Sources:** DIYMA 2013-11-29 · DIYMA 2021-12-14 · DIYMA 2017-07-30

In plain English: Distortion is sound a speaker adds that wasn't in the music, usually extra tones at multiples of the note being played. He says don't only look at how much there is; look at its type and its shape. Even-order distortion comes from something lopsided in the speaker, often the spider (the springy fabric ring that holds the voice coil centred), and he says it masks (covers up) detail. Odd-order distortion comes from the voice coil moving out of the magnet's gap on big movements, and he says it sounds mechanical. Distortion given as a percentage converts to decibels with 20 × log; he gives 0.2% as about −54 dB. Finally, close-up (near-field) measurements miss cone breakup, the point where the cone stops moving in one piece and starts to flex, so measure from far enough away.

5.2 #15 In cars off-axis behaviour matters more than on-axis.

> You always listen off axis near reflecting surfaces, so judge drivers at the angle you'll hear them; off-axis holes cost midrange definition. RARE
>
> **Sources:** Audiofrog blog Dec 2020 · Crutchfield

In plain English: "On-axis" means straight in front of a speaker, where it is usually measured. "Off-axis" means at an angle. In a car you almost never sit straight in front of a speaker, and the speakers sit near surfaces such as glass and the dashboard that reflect sound. So he says to judge a driver by how it performs at the angle you'll actually hear it from. An off-axis "hole" is a dip, a band

of notes that goes quiet at certain angles; he says such holes cost the midrange its definition (clarity). Many people judge speakers by their straight-ahead graph; in a car, he says, the angled response matters more.

5.2 #16 An in-car graph shows the car, not the speaker.

> A speaker outputs what its manual graph shows; every change at the seat is the car or a filter you added, so single-point in-car RTAs make every speaker look bad. Read them spatially averaged or smoothed.
>
> **Sources:** DIYMA 2021-08-06 · DIYMA 2015-12-22

In plain English: An RTA (real-time analyser) is a microphone plus software that shows how loud each frequency is where the microphone sits. His point: a speaker puts out what the graph in its manual shows. Any change you see at the seat comes from the car (its reflections and shape) or from a filter you added, not from the speaker. One microphone in one spot picks up a jagged pattern of peaks and dips from reflections, so it makes every speaker look bad. To read in-car graphs fairly, he says to use spatial averaging (measuring at several spots around the listening position and averaging them) or smoothing (a setting that evens out the fine jaggedness).

5.2 #17 A measurement is invalid only if you misread it.

> A calibrated mic is "correct" but the screen can mislead (comb filters, no distortion shown, no centre in stereo noise); the test is whether it correlates with hearing, like a scale that doesn't "feel" weight yet is still useful. RARE
>
> **Sources:** DIYMA 2017-06-28 · DIYMA 2017-10-28 · DIYMA 2021-07-09

In plain English: A calibrated measurement microphone (one whose own errors have been measured and corrected for) reports the sound at its tip correctly. He says the problem is how people read the screen. His examples: comb filters, a row of narrow dips caused by a sound and its reflection arriving at slightly different times; distortion, which the usual graph doesn't show; and stereo test noise, which has no centre, so it can't show the centre image. The real test is whether a measurement correlates with (lines up with) what you hear. His comparison is a weighing scale: it doesn't "feel" weight the way you do, yet it is still useful. Many people say measurements miss what ears hear; he says a measurement is only invalid if you misread it.

5.2 #18 Store sound boards and demo walls mislead.

> A board hurts every speaker mounted on it, a low-Q midbass sounds thin on a board but full in a door, and a mistuned board (wrong port tuning) hurts every brand on it.
>
> **Sources:** DIYMA 2015-11-15 · B2 @57:53

In plain English: Shops often show speakers on a sound board or demo wall: a panel of speakers you can switch between. He says these mislead. A board hurts every speaker mounted on it. A low-Q midbass (a woofer designed with less of a bass bump) sounds thin on a board but full once it is in a car door, because a door mounting adds bass that the board doesn't. And if the board itself is built wrong, for example a port (the tube in a bass box) tuned to the wrong frequency, every brand on it suffers. So a store demo tells you little about how a speaker will sound in your car.

5.2 #19 Liking is not proof: ask what was compared, and demand quantified claims.

> "I did all this and I like it" doesn't mean correct; unquantified workarounds become myths ("play with delays for width"). Test claimed DSP differences with a null test (align, invert, mix). RARE
>
> **Sources:** DIYMA 2015-11-14 · DIYMA 2021-08-14 · DIYMA 2021-01-04

In plain English: He says enjoying a result doesn't prove it is right. When someone says "I did all this and I like it", ask what they compared it with. "Quantified" means backed by numbers or measurements. Tips without numbers, such as "play with the delays to get more width", get repeated until they become myths. For claims that one DSP sounds different from another, he describes a null test, step by step: record the same music through both; line the two recordings up exactly in time; flip one upside down (invert its polarity); then mix them together. Whatever is identical cancels out, and only the real difference remains. He adds that working out what causes that difference is the harder part.

5.2 #20 Independent reviews and maker specs differ for honest reasons.

> Environments and techniques differ, so "gotcha" reviewing is silly, though some makers over-claim; he found an independent review useful for catching manual errors.
>
> **Source:** DIYMA 2017-08-09

In plain English: When an independent reviewer measures a speaker, the numbers often don't match the maker's spec sheet. He says there are honest reasons: different test environments and different measuring techniques give different results. So he calls the "gotcha" attitude, treating any mismatch as proof the maker lied, silly. He also admits that some makers do over-claim. And he found one independent review of his own product useful: it caught small mistakes in the owner's manual that he would fix.

5.2 #21 "Quality" has two meanings, and there is no perfect speaker.

> Engineers mean meeting spec consistently, buyers mean meeting expectations; change one thing in a speaker and "a thousand other things" change, so each is right only for an application.
>
> **Sources:** SQOLOGY Pt 1 @49:05 · CMA 2022 @7:02

In plain English: He says the word "quality" means different things to different people. To an engineer, a quality product is one that meets its specification every time, unit after unit. To a buyer, quality means it does what they expected. He also says there is no perfect speaker. Change one thing in a speaker's design and "a thousand other things" change with it, so every design is a set of trade-offs. That means a speaker is only right for a particular application: a particular job, location and use.

5.2 #22 Car drivers need car-grade qualification.

> Audiofrog's 40-page standard includes a 200-hour heat-humidity cycle (90 °C to −35 °C), vibration, thermal shock and UV; home drivers aren't built for this.
>
> **Sources:** DIYMA 2015-07-27 · CMA 2021 @46:02

In plain English: A car is a harsh place for a speaker: it bakes, freezes, gets damp and shakes. He says car speakers must pass car-industry testing. Audiofrog's standard runs to 40 pages. It includes a 200-hour cycle of heat and humidity between 90 °C (194 °F) and −35 °C (−31 °F), plus

vibration, thermal shock (sudden temperature changes) and UV (sunlight) exposure. He says drivers made for home speakers aren't built for this.

5.2 #23 Buy performance, not features; ask a brand what makes it better.

> Optical inputs, "super-duper" DACs and gold RCAs are features whose importance is "rarely substantiated"; a DSP's software is the hard part, and "12 channels" or "passion" is not an answer. RARE
>
> **Sources:** DIYMA 2008-06-11 · OFCAP 19 @1:30:03

In plain English: A feature is something a product has; performance is how well it does its job. He names features whose importance, he says, is "rarely substantiated" (rarely shown to matter): optical inputs (a digital connection by light cable), "super-duper" DACs (the chips that turn digital audio back into a signal) and gold-plated RCA plugs (the usual audio cable connectors). For a DSP, he says the software, the part you use to set it up and tune it, is the hard part to get right. So in practice: ask a brand what makes its product better. Answers like "12 channels" or "passion" don't answer that question. Many buyers compare products by their feature lists; he says to compare what they actually do.

5.2 #24 Run claims through the "Truth-O-Meter": what is the mechanism?

> Metaphor is not engineering ("effortless", "blacker blacks", "chocolaty" bass, tweeters "made of weasel noses"); he says audio has much "purposeful confusion". Debunking without a working alternative is no better than a falsehood. RARE
>
> **Sources:** DIYMA 2011-03-26 · DIYMA 2021-09-15 · B2 @17:53

In plain English: His "Truth-O-Meter" is a simple test for any audio claim: what is the mechanism, the actual physical reason it would work? He says metaphor is not engineering. He gives examples of language he rejects: "effortless" sound, "blacker blacks", "chocolaty" bass, and (mocking marketing) tweeters "made of weasel noses". He says audio has a lot of "purposeful confusion", meaning some of the confusion is created on purpose. He also warns debunkers: knocking down a false claim without offering something that actually works is no better than the false claim itself.

5.2 #25 A subjective impression is an experience, not data, but "unmeasured" isn't "imaginary".

> Only someone who knows every measuring tool can say which; trained ears can be an objective tool.
>
> **Source:** SQOLOGY Pt 1 @45:03

In plain English: A subjective impression is what a person hears and feels while listening. He says it is an experience, not data. But he warns against the opposite mistake too: something that hasn't been measured isn't automatically imaginary. Telling which is which, he says, takes someone who knows every measuring tool. He adds that trained ears can themselves be an objective tool.

5.2 #26 Read the books, not the forums.

> Toole's *Sound Reproduction* first, then Colloms and Darlington's *High Performance Loudspeakers*, Geddes and Voishvillo on nonlinearity, Griesinger's site and the ARTA near-field note; Facebook and YouTube "expertise" often treats car acoustics as unknowable.
>
> **Sources:** DIYMA 2008-10-01 · DIYMA 2021-12-24 · DIYMA 2021-07-26

In plain English: He sends learners to books and researchers rather than forums. His list: Toole's book Sound Reproduction first; then High Performance Loudspeakers by Colloms and Darlington; Geddes and Voishvillo on nonlinearity (the ways speakers stop behaving perfectly, which is what produces distortion); the website of Griesinger, an acoustics researcher; and the ARTA near-field note (a technical note on measuring speakers up close, for the ARTA measurement software). He says Facebook and YouTube "expertise" often treats car acoustics as unknowable.

5.2 #27 Forum culture: "the loudest idiot is the authority".

> Non-experts should qualify claims ("I think"), and answers boiled down too far stop being true.
>
> **Sources:** DIYMA 2008-07-27 · OFCAP 19 @17:03

In plain English: He describes forum culture as "the loudest idiot is the authority": whoever shouts loudest gets treated as the expert. He wants people who aren't experts to qualify their claims, with words like "I think". He also warns that answers boiled down too far, to make them short and simple, stop being true.

5.2 #28 Ignore claims of custom MS-8 firmware.

> It was an encrypted binary, only one tiny update ever shipped, and the only real mods were hardware (op-amps). RARE
>
> **Source:** DIYMA 2021-09-28

In plain English: This point is about one product: the JBL MS-8, a car-audio sound processor he often discusses. Firmware is the software built into a device. Some people claimed to have custom MS-8 firmware; he says to ignore those claims. The firmware went to the factory as an encrypted binary (scrambled, compiled code that can't simply be read or edited), and only one tiny update was ever released. The only real modifications people made were to the hardware, such as swapping op-amps (small amplifier chips in the signal path).

5.3 Planning and buying a system

What this part is about: Planning a system and spending money wisely: where money matters most (installation and tuning), how to upgrade a factory system in stages, which parts are worth paying for, and what to avoid buying.

5.3 #1 Money saved on installation is the worst money you'll ever save.

> Cars need far more fabrication, disassembly and troubleshooting than home audio; some customers have paid six stores.
>
> **Sources:** OFCAP 19 @2:28:05 · SQOLOGY Pt 2 @38:02

In plain English: Installation is the work of fitting the system into the car. He says money saved there is "the worst money you'll ever save". Setting up home audio mostly means placing speakers and plugging things in. A car needs far more: fabrication (building custom mounts, panels and boxes), taking the car's interior apart, and troubleshooting. He says some customers have been to six different stores and paid six times.

5.3 #2 A correct tune matters more than $200 vs $2,000 speakers; priorities are processing, installation, then brand.

> EQ is "THE ONLY THING" that overcomes the car's response; at a China dealer training every bad car's fault was installation, adjustment or design, never the gear. RARE
>
> **Sources:** OFCAP 19 @1:54:03 · DIYMA 2008-06-21 · DIYMA 2010-06-06

In plain English: "Tuning" means setting the processor's EQ, delays, crossovers and levels for your particular car. He says a correct tune matters more than whether the speakers cost $200 or $2,000. His order of priority is: signal processing first, installation second, the brand of gear third. EQ (equalisation: turning individual frequencies up or down) is, in his words, "THE ONLY THING" that overcomes the car's own effect on the sound, because even the flattest speaker stops being flat once it is mounted in a car. His evidence: at a dealer training in China, half the dealers planned to replace all their equipment because their cars sounded bad, but in every bad car the fault was installation, adjustment or system design, never the gear. Many people believe better speakers are the main route to better sound; he ranks the brand of gear last.

5.3 #3 Speakers still matter for output and distortion: a smaller budget should cost loudness, not accuracy.

> No DSP removes a bad speaker's distortion; his GS line keeps accuracy and gives up maximum output versus GB.
>
> **Sources:** DIYMA 2016-03-21 · OFCAP 19 @1:36:02

In plain English: This balances the previous point. Speakers still matter for how loud a system can play and how much it distorts. No DSP can remove the distortion a bad speaker makes. His rule for a smaller budget: it should cost you loudness, not accuracy. Audiofrog's GS line is built that way: compared with the GB line, it keeps the accuracy and gives up maximum output.

5.3 #4 Baseline first: full bandwidth with a sub, enough output, no distortion or rattles; then width and a stable centre.

> Width is his personal top priority; thick midbass is his pet peeve.
>
> **Sources:** OFCAP 38 @35:03 · OFCAP 19 @1:35:05

In plain English: He sets a baseline every system must reach before anything else. Full bandwidth with a sub: the system plays the whole range of notes, with a subwoofer for the deepest bass. Enough output: it plays loud enough. No distortion and no rattles. Only after that come the refinements: width (how far the sound spreads across the front of the car) and a stable centre (a singer who stays put in the middle). Width is his own top priority. "Thick midbass", too much of the upper-bass range, is his pet peeve.

5.3 #5 Decide early whether the system serves one seat or every seat.

> He would take an immersive GS system (rears plus upmixer) over a two-channel GB one; a big-budget system that makes the passenger ask "you paid how much?" is a lost sale. RARE
>
> **Sources:** OFCAP 38 @45:00 · CES 2016 @1:02

In plain English: Many high-end car systems are tuned for the driver's seat only. They use delays (holding back the nearer speakers) so that, from that one seat, each sound seems to come from the right place on an imaginary stage, at the expense of the other seats. He says to decide at the start whether the system is for one seat or for every seat. He would choose an immersive system built from Audiofrog's GS speakers, with rear speakers plus an upmixer (a processor that spreads two-channel stereo across more speakers), over a two-channel system built from the higher-end GB speakers. And if a big-budget system makes the passenger ask "you paid how much?", he counts that as a lost sale. Many people believe sound quality means optimising for the driver alone; he disagrees.

5.3 #6 Factory-system upgrade path: speakers first on the factory amp and EQ, then power, then sub, rears and DSP.

> Only about 10–20% of a factory EQ compensates for the stock speakers; example tiers on a Bose truck ran about $2,000 / $3,500 / $4,500 / $7,500. RARE
>
> **Sources:** CMA 2023 @11:00 · CMA 2023 @49:01

In plain English: Many cars come with a factory sound system that has its own amplifier and EQ. He describes upgrading it in stages. First, replace the speakers but keep the factory amp and EQ. Next, add power (aftermarket amps). Then add a subwoofer, rear speakers and a DSP. Many people believe new speakers are pointless without new amps and a DSP, because the factory EQ was tuned around the cheap stock speakers. He says only about 10–20% of a factory EQ makes up for the stock speakers; the rest corrects for the car's cabin, so it stays useful after a speaker swap. His example tiers on a Bose-equipped truck cost about $2,000, $3,500, $4,500 and $7,500.

5.3 #7 Keep good factory processing; don't "sum it back to flat".

> Factory all-pass filters can't be un-EQ'd, so on Logic 7 / Mark Levinson-type systems "DO NOT TRY TO SUM" the car back; in upmixing factory systems wire each factory channel one-to-one to new amps, fit stronger speakers, then just re-EQ and set levels. RARE
>
> **Sources:** CMA 2021 @35:00 · DIYMA 2021-09-26 · B1 @58:34

In plain English: Many premium factory systems split and process the sound heavily before it reaches each speaker. A common aftermarket approach is to "sum" (recombine) the factory channels back into a plain, flat stereo signal, then process it again with a new DSP. He says don't do this with good factory processing. One reason: factory systems use all-pass filters, which shift timing (phase) without changing loudness, and those can't be undone with EQ. So on Logic 7 or Mark Levinson-type systems, which use an upmixer (a processor that spreads stereo across many speakers), his words are "DO NOT TRY TO SUM" the car back. Instead, in an upmixing factory system, connect each factory channel one-to-one to a new amp channel, fit stronger speakers, then just re-EQ and set the levels. Many installers believe the first step is always to recover a flat signal; he disagrees for these systems.

5.3 #8 The DSP is your installer's tool: let them choose it.

> If you tune yourself, buy the one whose features you can learn; any amp "that doesn't suck" works, but for high-end systems prefer separate DSP and amps (5.2 #8).
>
> **Sources:** OFCAP 19 @2:46:08 · B1 @38:23

In plain English: He sees the DSP as the installer's tool, so let the installer choose the one they know, the way a carpenter picks their own tools. If you will tune the system yourself, buy the one whose features you can learn to use. For amplifiers, he says any amp "that doesn't suck" works. For high-end systems, though, he prefers a separate DSP and separate amps, for the gain reasons explained in 5.2 #8.

5.3 #9 Pay only for parts that buy something audible.

> With a sub you don't need an 8" midbass; tweeters close to the woofer can stay passive; swapping one good 6.5" for another at similar power changes little; most "voicing" you dislike is fixable with EQ. Don't reuse a passive crossover on a different speaker.
>
> **Sources:** CMA 2023 @7:01 · DIYMA 2021-12-03 · DIYMA 2021-12-21

In plain English: He says to spend money only where you'll hear the difference. His examples. With a subwoofer handling the deep bass, you don't need an 8-inch midbass (the woofer in the door). Tweeters mounted close to the woofer can stay on a passive crossover (the coils and capacitors supplied with the speakers) instead of each getting its own amp channel. Swapping one good 6.5-inch speaker for another at similar power changes little. "Voicing" is a speaker's built-in tonal balance; most voicing you dislike can be fixed with EQ. And don't reuse a passive crossover with a different speaker from the one it was designed for.

5.3 #10 Skip showpiece fabrication if sound is the goal.

> A beautiful trunk with cheap amps and speakers sounds poor; use factory locations when they work.
>
> **Sources:** OFCAP 38 @32:04 · OFCAP 38 @1:05:01

In plain English: Showpiece fabrication means show-car work, such as a beautifully built trunk display. He says that if good sound is your goal, skip it. A gorgeous trunk with cheap amps and speakers still sounds poor. And where the car's factory speaker locations work, use them.

5.3 #11 The core upgrade is still "better front plus real bass".

> Carmakers can't add a 40 lb sub box and a 60 A amp (his OEM limits: about 30 A and 12 lb), and factory head units can beat aftermarket ones (balanced outputs).
>
> **Sources:** OFCAP 19 @58:06 · DIYMA 2008-06-21

In plain English: The aftermarket means parts added after the car leaves the factory. He says the core thing it offers is still "better front plus real bass": better front speakers and a proper subwoofer. Carmakers can't add a 40 lb (about 18 kg) subwoofer box and an amp drawing 60 A. The limits he gives for factory (OEM) systems are about 30 A and 12 lb (about 5.4 kg). He also says a factory head unit (the radio unit in the dash) can be better than an aftermarket one; his example is balanced outputs, a type of signal connection that rejects electrical noise.

5.3 #12 No adjustable system is "entry-level".

> Two front speakers plus a DSP amp, installed, costs at least about US$2,500, so that buyer expects great sound and the shop must deliver it.
>
> **Source:** B2 @25:03

In plain English: "Entry-level" means the cheapest, most basic tier, where buyers expect little. He says no adjustable system (one with a DSP that can be tuned) is entry-level. Even two front speakers plus an amp with a built-in DSP, installed, costs at least about US$2,500. Someone spending that much expects great sound, so the shop has to deliver it.

5.3 #13 Cheap fixes outrank exotic ones.

> A clean windshield "probably" matters more than rear delay settings, quieter tyres were his best midbass upgrade, and 5.1 vs 7.1 is splitting hairs. RARE
>
> **Sources:** DIYMA 2012-02-02 · DIYMA 2016-01-25

In plain English: He says simple, cheap fixes often beat exotic ones. A clean windshield "probably" makes a bigger difference than the delay settings for the rear speakers. Quieter tyres were his best midbass upgrade. And arguing over 5.1 versus 7.1 (surround-sound layouts with five or seven main channels plus a sub) is splitting hairs. The source notes don't spell out his reasoning for the windshield or the tyres. Many people chase advanced settings and formats first; he points to the basics.

5.3 #14 Never buy used speakers, amps or DSPs online except from the maker.

> It's worse than used tyres; Audiofrog drivers can't be reconed, so "maybe you can have it fixed" means walk away, and pay with recourse if you must. RARE
>
> **Source:** DIYMA 2020-07-01

In plain English: He advises never buying used speakers, amplifiers or DSPs online unless they come from the manufacturer. He calls it worse than buying used tyres. Reconing means repairing a blown speaker by replacing its cone, coil and suspension, and Audiofrog drivers can't be reconed. So if a seller says "maybe you can have it fixed", walk away. If you must buy used anyway, pay with recourse, meaning a payment method that lets you get your money back if the item turns out to be faulty. Many people see used gear as a smart bargain; he disagrees.

5.4 Choosing and working with a shop or installer

What this part is about: How to choose a car-audio shop and work with it, and how Andy thinks a good shop should handle tuning, troubleshooting and its own staff. Some points speak to customers; others speak to shop owners and installers.

5.4 #1 Go to the best, most expensive shop in town and have it build a mid-level system.

> Paying about 30% more for "the guy who can build the space shuttle" beats asking a cheap shop to work above its level. Tell them your budget. RARE
>
> **Source:** OFCAP 19 @2:43:04

In plain English: Many people believe a modest budget means going to a modest, cheaper shop. He says the opposite. Go to the best shop in town, tell them honestly what you can spend, and have them build a mid-level system. You may pay about 30% more, but the work is done by someone capable of very ambitious builds ("the guy who can build the space shuttle"). Asking a cheap shop to work above its usual level doesn't make it better at the job. So in practice: pick the top shop, tell it your budget, and let it build to that.

5.4 #2 Specify the outcome, not the method.

> Forum recipes (pod angles, a named DSP, rears out of phase) can make your goal impossible; good shops make you choose performance or your method.
>
> **Source:** OFCAP 19 @2:48:07

In plain English: Tell the shop the result you want, not how to get it. Many enthusiasts arrive with instructions picked up on forums: angle the speaker pods this way, use this particular DSP, wire the rear speakers out of phase. (A DSP, or digital sound processor, is a small computer that adjusts the sound sent to each speaker. "Out of phase" means one speaker pushes air while another pulls.) He says such recipes can make the result you actually care about impossible. A good shop will make you choose: the performance you want, or your method. It can't promise both.

5.4 #3 Don't shop installation on price or take a consultation then buy online.

> Saving $80–340 leaves the tech unmotivated "by human nature"; good installers live on product plus labour margin. RARE
>
> **Sources:** SQOLOGY Pt 2 @45:05 · OFCAP 19 @2:32:00

In plain English: He warns against two common habits: choosing an installer just because they are the cheapest, and getting a shop's advice, then buying the equipment online and bringing it in to be fitted. Saving something like $80 to $340 this way backfires. The technician who then does the work is, "by human nature", less motivated, and in a hurry to get the car in and out. Good installers make their living from two things together: the profit (margin) on the products they sell, and the labour they charge. Many people see buying gear online as simply smart shopping; his point is that it takes away part of what pays for careful work.

5.4 #4 The best shop won't tune a car it didn't build.

> Tuning is the payoff of its whole process; if you built it yourself, buy a measurement kit and get coached.
>
> **Source:** OFCAP 19 @2:40:02

In plain English: "Tuning" means setting up the sound processor so the system sounds right in that particular car: the level of each speaker, its timing, its crossover (which range of notes each speaker plays) and its EQ (boosting or cutting particular ranges of notes). He says the best shops won't tune a car someone else installed. For such a shop, tuning is the payoff of its whole process, the step where all its earlier work comes together. If you installed your own system, his advice is to buy a measurement kit (a measurement microphone plus software that shows what the speakers are really doing in the car) and get someone to coach you through tuning it yourself.

5.4 #5 Tuning is part of the installation, so bill real time for it.

> "System tuning isn't really optimizing, it's troubleshooting"; shops show four-month builds that don't sound great and say "not bad for the 20 minutes I spent" on the tune. RARE
>
> **Sources:** ME column Aug 2017 · B2 @22:22 · OFCAP 19 @1:08:06

In plain English: Many people treat tuning as a quick finishing touch after the "real" work of installing. He says tuning is part of the installation itself, and a shop should spend, and bill for, real time on it. In his words, tuning "isn't really optimizing, it's troubleshooting": it is mostly finding and

fixing problems, not polishing a system that already works. He describes shops proudly showing him cars that took four months to build and still didn't sound great, with the installer saying "not bad for the 20 minutes I spent" on the tune. The build got months; the sound got minutes.

5.4 #6 A professional tune needs an agreed endpoint and a time estimate.

> Open-ended hourly tuning is like a painter who won't name the colour; a front-plus-sub system takes him 1–2 hours.
>
> **Sources:** DIYMA 2015-07-14 · UMI-1 video @6:06

In plain English: Before a professional starts tuning, he says, the tuner and the customer should agree what "finished" means and roughly how long it will take. Charging by the hour with no defined end is, in his comparison, like hiring a painter who won't tell you what colour your walls will be: you are paying without knowing what result you will get. For a sense of scale, he says tuning a system with front speakers plus a subwoofer (the large speaker that plays the deepest bass) takes him 1 to 2 hours.

5.4 #7 Judge a tuning process by whether it reaches "correct", then by speed and teachability.

> He teaches only repeatable processes aimed at retailers' predictable, timely work; open-ended experiments (months of dash removal, by-ear all-pass fiddling) have unknown value.
>
> **Sources:** OFCAP 19 @3:19:02 · DIYMA 2021-09-07 · DIYMA 2021-06-20

In plain English: When comparing ways to tune a car, he puts the questions in order. First: does the method actually get the system to a correct result? Only then: how fast is it, and can it be taught to other people? He teaches only repeatable methods, because shops need work that is predictable and finished on time. Open-ended experiments, such as spending months taking a dashboard apart, or fiddling with all-pass filters by ear, have in his view unknown value. (An all-pass filter is a processor setting that shifts the timing, or phase, of a range of notes without changing how loud they are.)

5.4 #8 There are no "tuning tricks", only a process: pick one and don't blend gurus.

> A barrel of tips from a contest winner isn't a process (he posted a 29-step parody of ear-tuning); if someone can't say why a step works, he doesn't know that it works. RARE
>
> **Sources:** DIYMA 2018-07-02 · OFCAP 19 @3:20:06 · DIYMA 2021-07-09

In plain English: Many hobbyists collect "tuning tricks" from contest winners and forum experts and mix them together. He says there are no tricks, only a process: a set of steps done in a sensible order, each for a known reason. A pile of tips, even from a competition winner, is not a process. To make the point, he once posted a 29-step parody of tuning by ear. His test for any step: if the person recommending it can't say why it works, they don't actually know that it works. So in practice: choose one teacher's process and follow it, rather than blending several "gurus".

5.4 #9 Measure first, listen last.

> He never listens until measurements say the car is ready, then listens once; hours of touch-up afterwards mean the measurement, the tuner or the process failed. Tuning purely by ear is aimless

> fiddling, and "golden ears" posturing scares customers. RARE
>
> **Sources:** DIYMA 2017-06-28 · DIYMA 2021-07-09 · DIYMA 2016-01-29

In plain English: Many people believe the best tuners do everything by ear. His order is the reverse. Measure first, with a microphone and analysis software, and don't listen at all until the measurements say the car is ready. Then listen once, as a final check. If hours of touching-up by ear follow, something has failed: the measurement, the person tuning, or the process. Tuning purely by ear, he says, is aimless fiddling. He also warns that tuners who pose as having "golden ears" (a special gift of hearing) scare customers rather than reassure them.

5.4 #10 More precision is not more performance for a professional.

> Beyond a point precision buys nothing; hobbyists may enjoy the endless search, professionals can't. Tips you can't explain are "monkeys", so experiment on your own car, not customers'.
>
> **Sources:** OFCAP 19 @1:53:04 · ME tag page 2017

In plain English: He says that beyond a certain point, measuring and adjusting more precisely brings no improvement in sound. In the source, his examples of precision that adds nothing are laser-aligning the tweeters (small speakers for the highest notes) and midrange speakers, and ultra-precise phase measurement between them. A hobbyist working on his own car may enjoy that endless search; a professional being paid by a customer can't afford it. He calls tips you can't explain "a barrel of monkeys likely to make a big mess." So in practice: try new ideas on your own car, not on customers' cars.

5.4 #11 Cars usually need fixing before tuning; troubleshoot before blaming the car.

> Only one car ever reached his office ready to tune. One contest car had no centre because the owner typed distances where the DSP wanted delays; long forum "breakthrough" sagas usually end when the delays are finally right. If you followed the manual and it didn't work, assume you made a mistake. RARE
>
> **Sources:** OFCAP 19 @1:58:00 · B1 @44:23 · B1 @47:54

In plain English: He says most cars brought in for tuning first need something fixed; only one car ever reached his office actually ready to tune. His examples here involve delays. Sound from the nearer speakers reaches you first, so a DSP can hold those speakers back by a few thousandths of a second (milliseconds) so that all the sound arrives together. One contest car had no "centre" (the singer didn't seem to stand in the middle) because the owner typed distances into boxes where the DSP expected delay times. He adds that long forum threads about a sudden "breakthrough" usually end when the person finally gets the delays right. So in practice: if you followed the manual and it didn't work, assume you made a mistake, and look for it before blaming the car.

5.4 #12 Blame setup before gear.

> About 70% of GB25 warranty claims are DSP input clipping (about 25% broken terminal screws); when a "blown speaker" appears suddenly, ask what changed in the source (new phone, new streaming app). RARE
>
> **Sources:** OFCAP 19 @1:49:05 · DIYMA 2022-03-13 · DIYMA 2022-01-15

In plain English: The GB25 is a tweeter made by Audiofrog, Andy's company. He says about 70% of GB25 warranty claims come down to clipping at the DSP's input, and about 25% to broken terminal screws (the screws the speaker wires attach to). "Clipping" happens when a signal is pushed past the most a device can handle: the tops of the sound wave get cut off, producing harsh distortion that can destroy tweeters. In other words, the most common cause was the system's setup, not the speaker. So in practice: when a speaker suddenly seems "blown", first ask what changed at the music source, such as a new phone or a new streaming app. He also suggests asking whether it happens with music stored somewhere else.

5.4 #13 Never "quick-fix" someone else's tune.

> You can't know their compromises; a contest "tweak booth" undid a car's noise compromise and cost more points than the noise.
>
> **Source:** DIYMA 2016-01-29

In plain English: Every tune contains compromises: choices that accept one small problem to avoid a bigger one, for reasons only the original tuner knows. He says never make quick "fixes" to someone else's tune, because you can't know what those compromises were. His example: before a contest, a car had a deliberate compromise over engine noise. Overnight, a "tweak and tune" booth at the event redid its wiring connections. The noise was still there, the left and right channels ended up reversed, and the connections were unreliable. That cost the car more points than the noise would have.

5.4 #14 Expertise is knowing the general case well enough to spot the outlier.

> You don't ask a heart surgeon whether he has operated on someone named Bruce; cars are "more alike than different", and knowing a DSP's menus isn't knowing audio. RARE
>
> **Sources:** DIYMA 2021-12-22 · DIYMA 2021-08-26

In plain English: Many people believe an expert is someone who has already worked on their exact car. His comparison: you wouldn't ask a heart surgeon whether he has ever operated on someone named Bruce. The name doesn't matter; what matters is knowing how hearts work. Cars, he says, are "more alike than different". Real expertise means understanding the general case so well that you can recognise the rare car that truly is different (the outlier). In the same way, knowing your way around one DSP's menus is not the same as understanding audio.

5.4 #15 "Correct" is the beginning of expertise; the rest is backing off correct for the customer.

> A shop serving only perfect spaghetti had better be in New York. His case: a 1970s-disco fan boosted bass 12 dB at the head unit, clipping it and blowing tweeters and midbass; the fix was a higher, pro-style crossover, giving up up-front bass he didn't care about. RARE
>
> **Sources:** B2 @42:16 · B2 @43:34

In plain English: He says learning to get a system technically "correct" is only the start of expertise. The rest is knowing how to step back from correct to please a customer who wants something different. His analogy: a restaurant that serves only perfect spaghetti had better be in a big city like New York, because elsewhere there aren't enough customers who want exactly that. His real case: a fan of 1970s disco kept blowing his speakers. Those recordings have little bass, so he turned the bass up by 12 dB at the head unit (the car radio). A dB, or decibel, is the unit for

loudness and signal level; 12 dB is a large boost. That drove the head unit into clipping (overload distortion) and destroyed the tweeters and the midbass speakers (the door speakers that play upper bass). A crossover is the dividing line that decides which speaker plays which range of notes. The fix was a higher crossover, set up the way professional sound systems are, giving up bass from the front speakers, which this customer didn't care about.

5.4 #16 If good sound depends on techs who love audio, "we suck".

> A written process lets people who don't love audio deliver a passable car; otherwise assign one person to check every car. Never invent on the critical path; schedule skill-building.
>
> **Sources:** OFCAP 19 @2:03:05 · OFCAP 38 @24:00

In plain English: If a shop's good sound depends on having technicians who happen to love audio, his blunt verdict is "we suck". A written, step-by-step process means that technicians who don't especially love audio can still deliver a car that sounds passable. Without such a process, a shop should assign one person to check every car. "Critical path" is a planning term for the chain of tasks that decides when a job is finished. He says never invent new methods on that path, that is, on work that is holding up delivery; instead, schedule separate time for building skills.

5.4 #17 Shops that won't charge enough or pay enough create unhappy customers.

> Installer ads offer about $35K, less than fast food; installation is now a tech job that needs trained leads.
>
> **Source:** OFCAP 19 @2:10:02

In plain English: He links low prices and low wages to unhappy customers. Job ads for installers offer about $35,000 a year, which he says is less than fast food pays. Yet installing audio in modern cars is now a technical job, and shops need trained lead technicians to do it. His point is that a shop that won't charge enough, or pay enough, ends up with unhappy customers.

5.4 #18 Use binaural recordings to demo upgrade tiers and document builds.

> Record each stage with a dummy-head mic; customers hear on headphones whether bass is up front and there's a centre (not loudness). RARE
>
> **Sources:** CMA 2023 @49:01 · Audiofrog Binaural (COMPANY)

In plain English: A binaural recording is made with a "dummy head": a model of a human head with a microphone in each ear. Played back on headphones, it sounds much as if you were sitting where the head was. He suggests recording the car at each stage of an upgrade. Customers can then hear on headphones the things that matter, such as whether the bass seems to come from the front and whether there is a clear centre (a singer who seems to stand in the middle), rather than just which version is louder. The same recordings also document each build.

5.5 Selling and the business of car audio

What this part is about: How Andy thinks car audio should be sold to customers, and how he sees the business side of the industry: shop margins, online sales, factory-fitted systems, training, product design and where car audio is heading.

5.5 #1 Customers want reassurance, not an education.

> He doesn't want to learn dog surgery, he wants a vet he trusts; explain only enough to show their risk is covered. (Host Matt Schaeffer argued for educating clients.)
>
> **Sources:** OFCAP 19 @40:01 · OFCAP 19 @49:01

In plain English: Many people in the trade believe the way to sell is to teach the customer all about the technology. He disagrees: customers mainly want to feel they are in safe hands. His comparison: if his dog needs surgery, he doesn't want to learn how to do it; he wants a vet he trusts. So a salesperson should explain only enough to show the customer that their risk is covered. The podcast host, Matt Schaeffer, argued the other side: that shops should educate their clients.

5.5 #2 Qualify customers in their own car by observation.

> Turn their stereo to about three-quarters (if they never listen that loud they'll turn it down) and play what they drove in with; your role is "investigator and translator". RARE
>
> **Source:** OFCAP 19 @1:01:05

In plain English: "Qualifying" a customer is sales language for finding out what they really want and need. He says do it by watching them in their own car, not just by asking questions, because customers won't honestly say how loud they listen. With the customer in the driver's seat, turn their stereo to about three-quarters volume and play the music they arrived with. If they never listen that loud, they'll reach over and turn it down. The salesperson's job is to be an "investigator and translator": find out what the customer wants, then translate it into what can be done for their budget.

5.5 #3 Find out whether imaging matters before selling it, and don't treat club listeners as lesser.

> Ask where music last moved them; concert-goers, club-goers and home-system lovers want different presentations. Club listeners still need a DSP, because impact comes from correct midbass-to-sub phase and DSPs keep drivers safe. RARE
>
> **Sources:** OFCAP 19 @1:19:00 · B2 @43:02 · B2 @46:57

In plain English: "Imaging" is the illusion that singers and instruments are placed at particular spots in front of you, as on a stage. Many sound-quality fans assume every customer wants this. He says find out first, by asking where music last moved them. People who love live concerts, people who love nightclubs, and people who mainly enjoy their own home hi-fi each want music presented differently. Club listeners are not lesser customers, and they still need a DSP. In his view, the punch they want comes from the midbass speakers and the subwoofer being correctly in phase (moving together), and the DSP also keeps the speakers ("drivers") safe from damage.

5.5 #4 Demo late, as confirmation; never open with "what's your budget?"

> The best demo proves you heard the customer's story; read the car they arrived in and test budget by showing something and hinting at price.
>
> **Sources:** OFCAP 19 @1:22:03 · OFCAP 19 @1:26:01

In plain English: A "demo" is letting the customer hear a display car or system. He says don't start with one, and never open the conversation with "what's your budget?". Listen to the customer first. The best demo comes near the end and proves you heard their story: it confirms you understood what they want. To get a feel for their budget, look at the car they arrived in, and show them something while hinting at its price to see how they react.

5.5 #5 Selling is persuading someone to trust you; scripted display-to-display selling fails.

> Untrained salespeople lose customers who made the effort to find a specialist.
>
> **Source:** OFCAP 19 @1:28:07

In plain English: He sees selling as persuading someone to trust you, not walking them through a script from one display to the next. Customers who went to the trouble of finding a specialist shop are lost when they meet a salesperson who hasn't been trained.

5.5 #6 With modern cars, assess constraints before pitching; "rip it all out" loses.

> A modern car may have 32 speakers designed by PhDs and a radio and amp that can't come out; sell what you can change. RARE
>
> **Source:** OFCAP 19 @55:03

In plain English: A common pitch is to tell the customer the factory system is junk and must all be replaced. He says that pitch loses with modern cars. A modern car may have 32 speakers in a system designed by engineers with PhDs, and its radio and amplifier often can't be removed. So in practice: first work out what can and can't be changed in this particular car, then sell improvements to the parts you can change.

5.5 #7 Answer "what's your cheapest option?" by explaining why you aren't the cheapest.

> Cite equipment, care for the car, reputation and expertise; you plant a seed for a friend who isn't price-shopping. RARE
>
> **Source:** MEA KnowledgeFest clip

In plain English: When someone phones asking only for the cheapest option, he suggests explaining why your shop isn't the cheapest: the equipment it has invested in, the care it takes with customers' cars, its reputation, and its expertise. That caller may still go elsewhere. But you have planted a seed: they may later mention you to a friend who isn't shopping on price.

5.5 #8 Explain labour charges; customers arrive defensive

> after being gouged elsewhere.
>
> **Source:** OFCAP 19 @2:38:02

In plain English: He says shops should explain what their labour charges are for, because many customers arrive on guard after being overcharged ("gouged") somewhere else.

5.5 #9 "Nobody's asking for that" can be an opportunity

> that just means "that would take work".
>
> **Source:** OFCAP 19 @2:22:04

In plain English: When people in a shop say "nobody's asking for that", he suggests it may really mean "that would take work". He adds the other side: with a little work, you could be the only shop in town offering it, and charge accordingly.

5.5 #10 Honesty sells: say what you'll do, do it, and send buyers elsewhere when you don't fit.

> He would rather lose a sale than disappoint, recommends other brands, and says the truth is "a damn good story" for salespeople.
>
> **Sources:** OFCAP 19 @2:51:01 · DIYMA 2017-08-09 · B2 @29:47

In plain English: His sales philosophy is plain honesty. Tell the customer what you will do, then do it. If your product or shop isn't the right fit, send them somewhere that is. He would rather lose a sale than leave someone disappointed, and he recommends other brands when they suit a buyer better. He tells salespeople that the truth is "a damn good story": they don't need to exaggerate.

5.5 #11 Demos sell, but shops demo the wrong cars.

> Boom and competition cars don't speak to ordinary buyers who want a great $2k system.
>
> **Sources:** DIYMA 2008-12-02 · ME Audiofrog tag page

In plain English: He agrees that demo cars sell systems, but says shops often show the wrong ones. "Boom" cars (built for extreme bass and loudness) and competition cars don't speak to an ordinary buyer, who wants a great system for about $2,000. His point is that the demo should match what ordinary buyers actually want.

5.5 #12 Online selling and margin-cutting hollow out the category; software and tuning are worth paying for.

> He told Amazon that listing Audiofrog would kill it; no retail margin means no R&D money. Expecting free software is like expecting free doctor visits. RARE
>
> **Sources:** OFCAP 19 @2:32:00 · DIYMA 2008-07-05 · DIYMA 2017-07-25

In plain English: "Margin" is the difference between what a shop pays for a product and what it sells it for: the money that keeps the business running. He argues that selling online and cutting prices drains that money out of the whole high-end car-audio category. He told Amazon that listing Audiofrog products there would kill the brand, because with no retail margin there is no money for R&D (research and development of new products). He also says software and tuning are worth paying for: expecting free software is like expecting free visits to the doctor.

5.5 #13 How big brands lost the high end, and why Audiofrog started when it did.

> In the Great Recession brands chased emerging markets with cheap speakers; as money flowed back to developed markets, high-end car audio became an enthusiast opening that still needs installers.
>
> **Sources:** CMA 2021 @7:03 · SQOLOGY Pt 2 @26:05 · B2 @7:27

In plain English: This is his account of recent industry history. During the Great Recession (the worldwide economic downturn that began around 2008), big car-audio brands chased fast-growing markets in developing countries with cheap speakers. As money later flowed back to

wealthier countries, the high end had been left open: an opportunity for products aimed at enthusiasts. That, in his telling, is why Audiofrog started when it did. He adds that this market still needs installers.

5.5 #14 Premium buyers are about 35–70, not 16–24, and often decide with a partner.

> In about 25 years he never sold a premium system to a 16–24-year-old (except with dad paying); hence a friendly, self-effacing name that's memorable, lowers expectations and wins the household negotiation. RARE
>
> **Sources:** B2 @10:48 · B2 @11:36 · SQOLOGY Pt 2 @30:00

In plain English: Many people picture the typical car-audio buyer as a teenager or young adult. He says buyers of premium systems are about 35 to 70 years old, and they often decide together with a partner. In about 25 years he never sold a premium system to a 16- to 24-year-old, except when dad was paying. That is why he chose a friendly, slightly self-effacing (modest) brand name, "Audiofrog". It is memorable, it doesn't promise too much (the way a name like "perfect speaker" would), and it doesn't put off the partner who has a say in the purchase.

5.5 #15 Three kinds of shops sell audio, and all must deliver great sound.

> Audio specialists, accessory shops that diversified, and high-end upfitters/restorers; techs must master woodwork, electronics and disassembly too.
>
> **Source:** B2 @20:22

In plain English: He describes three kinds of businesses that sell car audio: dedicated audio specialists; accessory shops that branched out into other products and services; and high-end upfitters and restorers (businesses that customise or restore vehicles). All three must deliver great sound that the customer loves. Their technicians need to master woodwork, electronics and taking cars apart, as well as audio.

5.5 #16 Factory systems are engineered by experts; learn what they do before "improving" them.

> OEM teams of PhDs spend years per system; aftermarket listeners judge them only on bass and loudness and miss the envelopment and two-seat imaging already there. Many carmakers deliberately image in front of each seat. RARE
>
> **Sources:** SQOLOGY Pt 1 @59:05 · B2 @37:25 · B1 @56:27

In plain English: "OEM" (original equipment manufacturer) here means the carmaker and the system it fits at the factory; "aftermarket" means equipment bought and installed later. Many enthusiasts assume factory systems are poor. He says they are designed by teams of PhD engineers who spend years on each system. Aftermarket listeners tend to judge them only on bass and loudness, and so miss what is already there: "envelopment" (the feeling of being surrounded by sound) and imaging that works for two seats, not just the driver's. Many carmakers deliberately make the sound image appear in front of each seat. So in practice: learn what a factory system does before trying to "improve" it.

5.5 #17 Good OEM premium systems beat typical aftermarket installs, and factory Harman drivers outperform many aftermarket ones.

> BMW owners complain about tuning, bass and format, not the drivers; Gary Biggs's IASCA BMW kept every factory driver. RARE
>
> **Sources:** DIYMA 2008-07-09 · DIYMA 2022-01-10

In plain English: A "driver" is a single speaker unit. Harman is a large audio company that supplies factory sound systems to many carmakers. He says good factory "premium" systems beat typical aftermarket installations, and the speakers Harman supplies to carmakers outperform many aftermarket speakers. BMW owners, he notes, complain about the tuning, the bass and the format, not the speakers themselves. (In the sources, "format" covers complaints such as how the Logic 7 surround processing sounds, where the centre image sits, and quiet rear speakers in surround mode.) His example: Gary Biggs's BMW, built for IASCA sound-quality competition, kept every factory speaker.

5.5 #18 The aftermarket isn't obsolete: OEM and aftermarket do different jobs.

> Carmakers work within weight, electrical and reliability limits and buy audio brands for the badge; enthusiasts who disagree with OEM tuning keep the aftermarket alive.
>
> **Sources:** Big 5 / CE Outlook Jul 2018 · DIYMA 2021-04-15

In plain English: Many people believe good factory systems have made aftermarket car audio obsolete. He says the two do different jobs. Carmakers must work within limits on weight, on the car's electrical system and on reliability, and they pay for well-known audio brand names mainly for the badge. Enthusiasts who disagree with how the factory tuned their system are what keep the aftermarket alive.

5.5 #19 Self-tuning DSPs exist because the industry can't train enough tuners.

> Harman wouldn't fund 30 trainers, so the MS-8 tuned itself; 473 sold in under two weeks, only 25 to DIYers.
>
> **Sources:** DIYMA 2011-06-09 · SQOLOGY Pt 1 @23:06

In plain English: This point is about one product, the JBL MS-8, a sound processor from Harman that could tune itself automatically using a microphone. He explains that self-tuning processors exist because the industry can't train enough people to tune cars by hand. Harman wouldn't pay for 30 trainers to teach installers, so the MS-8 was designed to tune itself. In under two weeks 473 were sold, and only 25 of those went to DIYers (people installing in their own cars).

5.5 #20 Products are shaped by specs and product managers; that's why DSPs lack upmixers.

> Head units lack good EQ because nobody writes it into the spec. Upmixer licences aren't expensive; they're missing because product managers don't understand or fight for them in finance meetings, and someone must then teach their use. RARE
>
> **Sources:** DIYMA 2016-06-03 · DIYMA 2017-07-07 · DIYMA 2021-08-01

In plain English: A "spec" (specification) is the document that says what a product must do; a product manager decides what goes into it. An "upmixer" is processing that takes ordinary two-channel stereo and creates extra channels from it, such as a centre or rear channel. EQ

(equalisation) adjusts the loudness of different ranges of notes. He says head units (car radios) lack good EQ simply because nobody writes it into the spec. In the same way, DSPs lack upmixers, not because the licences are expensive (they aren't), but because product managers don't understand them or don't fight for them in budget meetings. And once an upmixer is included, someone also has to teach people how to use it.

5.5 #21 Training separates shops, and Audiofrog's second job is teaching.

> It went dealer-direct, capped Audiofrog U classes at four techs, and uses the UMI-1 kit, blog and support to explain great audio; a measurement tool need not be expensive.
>
> **Sources:** GridOto Aug 2019 · B2 @18:41

In plain English: He sees training as what separates good shops from the rest, and teaching as Audiofrog's second job alongside making products. Audiofrog sells directly to its dealers (shops), limits its "Audiofrog U" training classes to four technicians at a time, and uses its UMI-1 measurement kit, its blog and its customer support to explain how to get great sound. He adds that a measurement tool doesn't have to be expensive.

5.5 #22 Standardize the tools you teach with.

> Dealers given a $150 parts list substituted parts to save $8, so Audiofrog sold one kit; about 1,500 sold by March 2021 (about 60% to DIYers).
>
> **Sources:** UMI-1 video @3:00 · CMA 2021 @13:05

In plain English: For years he gave dealers a parts list, about $150 worth of sound card and microphone, so they could put together their own measurement setup. They kept swapping in other parts to save $8, so he ended up supporting endless different combinations. So Audiofrog began selling one standard kit. About 1,500 had sold by March 2021, about 60% of them to DIYers. His lesson: if you teach a method, make sure everyone uses the same tools.

5.5 #23 Design shows care, and good products work even for people who don't read the manual.

> Performance first, then "wrap something beautiful" around it; the GB25 was designed so any small pod works. A business plan isn't what will happen, but it tells you how far off plan you are.
>
> **Sources:** B1 @16:38 · DIYMA 2017-07-02 · B2 @10:25

In plain English: His product philosophy: get the performance right first, then "wrap something beautiful" around it, because good design shows care. Good products should also work for people who never read the manual. For example, Audiofrog's GB25 tweeter was designed to work in any small pod (the small housing an installer builds to hold a tweeter). On planning, he says a business plan isn't a prediction of what will happen, but it does tell you how far off plan you are.

5.5 #24 Price-fixing talk in trade groups is illegal.

> He warned a trade-association group about Sherman Antitrust Act exposure.
>
> **Source:** OFCAP 19 @3:51:06

In plain English: Price-fixing is when competing businesses agree among themselves what prices to charge. He warned the admins of an industry trade-association group (a Facebook group, according to the sources) that a discussion about setting prices could expose its members to legal

liability under the Sherman Antitrust Act, the main US law against such agreements.

5.5 #25 The future is multi-seat and immersive.

> Carmakers are building audio zones, Atmos and spatial audio show where listening is going, and 30 years of two-channel SQ "leaves room for something else". Car DSP itself began in the early 1990s (Sony Mobile ES, Rockford Symmetry).
>
> **Sources:** CMA 2023 @39:03 · D&F live @26:03 · DIYMA 2021-10-15

In plain English: He sees car audio heading towards sound designed for every seat and towards "immersive" sound that surrounds the listener. Carmakers are building "audio zones" (different sound for different seats), and Dolby Atmos and "spatial audio" (formats that place sounds all around, and even above, the listener) show where listening is going. "SQ" means sound quality. After 30 years of two-channel (stereo) SQ, he says, there is "room for something else". On history, he notes that sound processing (DSP) in cars began in the early 1990s, with products such as Sony's Mobile ES line and Rockford's Symmetry.

5.6 SQ competitions and judging

What this part is about: Car-audio sound-quality ("SQ") competitions, such as those run by IASCA and MECA, where judges listen to cars and score them. Andy explains what winning does and doesn't prove, how cars get built to score points, and how he thinks judging should change.

5.6 #1 Winning IASCA or MECA "means you win contests", not that the car is correct.

> Some of the worst-sounding cars he heard were winners, and winning creates no new physics. Competition is "Formula 1", a contrived pursuit separate from the "luxury sedan" mainstream. RARE
>
> **Sources:** DIYMA 2013-03-20 · DIYMA 2021-09-09 · DIYMA 2020-10-07

In plain English: IASCA and MECA are organisations that run car-audio competitions. Many people assume a contest-winning car shows how a system should be done. He says winning just "means you win contests", not that the car is technically correct. Some of the worst-sounding cars he has heard were winners, and a win doesn't create new physics: a method isn't proven right just because a car using it won. He compares competition to Formula 1 racing: a contrived pursuit, separate from the "luxury sedan" mainstream, meaning the everyday systems most people listen to.

5.6 #2 How judged cars game the score sheet.

> Separate competition tunes per track, an all-pass filter for the centre-vocal track, deliberately out-of-phase treble that judges score as width, and uncorrelated pink noise to game the RTA lane; "it isn't great audio". His tongue-in-cheek 2012 recipe: build to the rules and talk the talk. RARE
>
> **Sources:** OFCAP 38 @54:01 · DIYMA 2022-01-31 · DIYMA 2012-02-05

In plain English: Judges score cars on a score sheet while playing a disc of test tracks. He describes four ways cars are built to score rather than to sound right. First, a separate competition tune for each judging track. Second, an all-pass filter (a timing, or phase, adjustment) used just for the track that tests whether the singer is centred. Third, treble deliberately set out of phase between left and right, which sounds spacious and which judges score as "width". Fourth,

uncorrelated pink noise used to game the part of the scoring done with an RTA. Pink noise is a hiss-like test sound containing all frequencies; "uncorrelated" means the left and right channels play different noise rather than the same noise. An RTA (real-time analyser) is a meter that shows how loud each range of notes is. His verdict: "it isn't great audio". In 2012 he posted a tongue-in-cheek "recipe" for winning: build to the rules and talk the talk. (In that same post he said the cars being discussed did sound great.)

5.6 #3 Install-point rules are often arbitrary.

> Zip ties every 18", fuses that protect nothing, wire gauges that "make no sense"; he won't run 4-gauge to a 100 W amp for points and moved to SQC-style events.
>
> **Sources:** DIYMA 2008-06-09 · DIYMA 2010-04-07

In plain English: Many competitions also award points for the installation, not just the sound. He says many of those rules are arbitrary: a zip tie every 18 inches (about 46 cm), fuses that don't actually protect anything, and wire sizes that "make no sense". He won't run 4-gauge wire (very thick power cable) to an amplifier of only 100 watts just to earn points. He also recalls an install judge marking him down because, the judge said, his amplifier wiring "couldn't work". He moved to SQC-style events instead; the sources don't describe that format in detail.

5.6 #4 Contest folklore without reasons, and secrecy, are red flags.

> No one who removes the steering wheel "for reflections" could name the frequency; centring the image with 31 sine waves is "really stupid"; hobbyists share while competitors "learn, conceal and win".
>
> **Sources:** DIYMA 2009-07-10 · DIYMA 2010-07-13 · DIYMA 2020-07-08

In plain English: He treats contest habits that come without reasons, and secrecy, as warning signs. Competitors "love to talk" about removing the steering wheel to get rid of a sound reflection, yet none could tell him what frequency the problem was at, or why. Centring the image (making the singer appear in the middle) using 31 sine waves, which are pure single-pitch test tones, he calls "really stupid". Elsewhere he explains that pure tones land in uneven dips caused by in-car reflections, and those dips differ between left and right. Finally, hobbyists share what they learn, while competitors "learn, conceal and win".

5.6 #5 Competitions don't drive innovation (2008).

> He knows "a hundred real scientists" building new tools, and none are judges; pitches that add complexity for points are suspect.
>
> **Source:** DIYMA 2008-06-11

In plain English: Many people believe competitions push the technology forward, the way motor racing is said to improve road cars. In 2008 he said they don't. He knew "a hundred real scientists" developing new audio tools, and none of them were judges. So a sales pitch that adds complexity just to win contest points should make you suspicious. This is his view as of 2008.

5.6 #6 Richard Clark's era: credit and critique.

> It moved car audio out of "witchery" and gave us differential inputs and low-impedance outputs, but "Time Misalignment" led competitors to dismiss delay entirely. Acknowledge the grey.
>
> **Sources:** DIYMA 2009-11-11 · DIYMA 2011-11-25

In plain English: Richard Clark is a famous name from car-audio competition's early decades; the sources also name Dave Navone and the AutoSound 2000 articles from that era. He gives that era credit: it moved car audio away from "witchery" (mysterious, unexplained claims) towards engineering. It also gave us differential inputs (amplifier inputs designed to reject electrical noise picked up in the car) and low-impedance outputs on head units (outputs that drive signal cables more robustly). But he also criticises it. An article called "Time Misalignment" was, in his view, technically right on a narrow point (that time alignment doesn't move the apparent source of a sound), yet it led many competitors to dismiss delay (time alignment) entirely. His conclusion: acknowledge the grey. The era did real good and some harm.

5.6 #7 Judging music and test discs push real listeners away.

> Free test tracks "suck", judges complain bass masks ambience "and they wonder why no hip-hoppers" enter, and the seven-drum imaging track can't check midbass placement.
>
> **Sources:** DIYMA 2011-10-15 · UMI-1 video @28:00

In plain English: Competitions judge cars using particular discs of music and test tracks. He says this pushes real listeners away. The free test tracks, in his words, "suck". Judges complain when bass masks the ambience (the subtle sense of the space a recording was made in), "and they wonder why no hip-hoppers" enter. And the seven-drum track used to judge imaging (where sounds appear across the stage) can't check where the midbass seems to come from. Midbass is the upper bass, usually played by the door speakers.

5.6 #8 Judges who mark down a strong centre are wrong, but play to the sheet.

> He advised MS-8 competitors in one-seat judging to switch Logic 7 off so the car sounds like the two-channel car judges expect.
>
> **Source:** DIYMA 2013-07-23

In plain English: This point relates to a specific product, the JBL MS-8 processor. Logic 7 is Harman's upmixer, built into the MS-8: it creates a centre channel and surround effects from ordinary stereo. Some judges mark down a car with a strong centre image (a solid, focused singer in the middle). He says those judges are wrong. But he is practical: if you compete, play to the score sheet. He advised MS-8 owners entering contests judged from one seat to switch Logic 7 off, so the car sounds like the plain two-channel car the judges expect.

5.6 #9 Score sheets are worthwhile but should reward two seats.

> No car should score a 10 until it sounds larger than itself and the same in every seat; he offered to help write two-seat rules.
>
> **Sources:** SQOLOGY Pt 1 @48:03 · SQOLOGY Pt 2 @21:02

In plain English: He supports having score sheets, but thinks they should reward cars that sound good from both front seats, not just the driver's seat. In his view, no car should get a top score of 10 until it sounds bigger than the car itself and sounds the same in every seat. He offered to help write judging rules for two seats.

5.6 #10 "Experiment with delays for depth" is wrong, and contest lasers are mostly about symmetry.

> Depth must be in the recording; laser-positioning a judge's head is irrelevant to daily driving.
>
> **Sources:** OFCAP 19 @3:23:07 · OFCAP 19 @3:38:00

In plain English: "Depth" is the sense that some performers stand farther back on the stage than others. Some people advise experimenting with the processor's delay settings to create depth. He says that's wrong: depth has to be in the recording itself. Some competition cars use lasers to position the judge's head precisely in the seat; he says that is irrelevant to everyday driving. When the podcast host said lasers in competition builds are really used to make the installation symmetrical, he agreed.

5.6 #11 He competed to show judges the multichannel future.

> An 18-speaker Logic 7 Mini (around 2004) and a Volvo S60R with room synthesis won shows and finals trophies; at CES, EMMA judges needed an hour and a briefcase of CDs to "break" Logic 7. Even Gary Summers' superb one-seat car sounded better to him in multichannel. RARE
>
> **Sources:** DIYMA 2021-09-08 · DIYMA 2016-02-05

In plain English: "Multichannel" means using more than the two stereo channels, for example adding a centre channel and side or rear channels, often created by an upmixer such as Logic 7. He entered competitions to show judges where he believed car audio was heading. He competed with an 18-speaker Mini using Logic 7 (around 2004) and a Volvo S60R using "room synthesis" (processing that creates a sense of space beyond the car). Both won shows and trophies at finals. At CES (the big annual consumer electronics show), judges from EMMA, another competition organisation, needed an hour and a briefcase full of CDs to find a track that "broke" Logic 7, that is, made it sound wrong. Even Gary Summers' superb competition car, set up for one seat (he called its stereo mode maybe the best one-seat car he had heard), sounded better to him in multichannel.

5.7 Recordings, listening and hearing

What this part is about: How records are actually made, what a well-tuned car should sound like, and how to listen to and judge a system. It also covers how our hearing works and where it falls short, including age, head movement and the tools people use to measure sound.

5.7 #1 Recordings are constructed fiction; aim for "believability", not "accuracy".

> Most records are assembled from parts recorded on different days ("a deep fake" of an event); early reflections are how our ears confirm what our eyes see.
>
> **Sources:** CMA 2023 @35:05 · OFCAP 19 @3:28:03 · OFCAP 38 @39:05

In plain English: Many people judge a system by how "accurately" it reproduces a recording. He says audiophiles tend to assume every recording places the listener in the audience, facing a stage. His point is that most records never were one real performance. The musicians are often recorded separately, on different days, and the parts are put together afterwards. He calls the result "a deep fake" of an event, so there is no single original moment to be accurate to. Instead he aims for believability: the sound should convince you of where the players are and what the space around them sounds like. Early reflections are the first echoes of a sound bouncing off nearby surfaces, arriving a split second after the sound itself. He says these are how our ears confirm what our eyes see, which he notes is why audiophiles close their eyes when they listen.

5.7 #2 The goal is a pleasing room, not no room.

> Stereo in an anechoic chamber sounded horrible to him; music is written for performance spaces (he recommends David Byrne's *How Music Works*), a car is too small and reflective to be reverberant, and a car can at best suggest a space larger than itself, even by adding a virtual room.
>
> RARE
>
> **Sources:** DIYMA 2017-07-05 · DIYMA 2017-07-04 · CMA 2023 @33:01

In plain English: An anechoic chamber is a special room lined with sound-absorbing material, so nothing bounces back: no echoes, no room sound at all. Many people believe the ideal is to get rid of the car's reflections so you hear only the speakers. He says he has heard a stereo pair in an anechoic chamber, and it sounded horrible. His point is that the room is part of the music. Music is written for the space it is played in; his examples are rock for clubs, and choral or baroque music for churches. So the aim is a pleasing room, the way concert halls are designed, not no room. "Reverberant" describes a space where sound keeps bouncing around and slowly dies away, like a church or a concert hall. He says a car is too small and reflective for that. Even with a virtual (electronically created) room added by a processor, the most a car can do is suggest a space bigger than itself.

5.7 #3 What a stereo that "works" sounds like.

> Instruments spread across the dash as panned, vocals centred, room and depth if recorded, true timbre, and speakers that "disappear"; that moment isn't magic, it happens whenever a car is tuned correctly. Stereo was built for one centred listener.
>
> **Sources:** OFCAP 19 @1:51:00 · B2 @39:51

In plain English: This is his description of the result you are aiming for. "Panned" means the mixing engineer placed each instrument somewhere between the left and right speakers. When the system works, you hear the instruments laid out across the dashboard where the engineer put them, with a centre-mixed singer in the middle. If the recording captured a room and depth, you hear those too. "Timbre" is the character of a sound, what makes a bassoon sound like a bassoon. And you stop hearing the speakers as the sources of the sound: they "disappear". Many people treat that moment as rare magic, reached only by ever more precise tuning. He says he gets it every time a car is tuned correctly. He also reminds us that stereo was designed for one listener sitting in the centre, between the two speakers. In a car you can't sit in the middle, which is why a car needs extra tuning steps to get there.

5.7 #4 Images can only land where the mix and the speakers allow.

> Engineers place sources mostly by level (about 6 dB moves an image halfway to one side); a sound from one speaker comes from that speaker, side reflections can't widen a stage beyond the car, and panning the balance fully left shows your stage's true edge. RARE
>
> **Sources:** EDUCAR · DIYMA 2017-08-21 · DIYMA 2017-08-18

In plain English: An "image" is the spot a sound seems to come from, even when there is no speaker there, like a singer who seems to stand in the middle of the dash. A dB (decibel) is the unit for differences in loudness. His point is that recording engineers mostly place sounds left or right by making them louder in one channel than the other. By his rule of thumb, about 6 dB more in one channel moves the image roughly halfway from the centre to that side. It follows that a sound played by only one speaker comes from that speaker. His thought experiment: put two speakers in the middle of a football field, with nothing for sound to bounce off, and turn the balance fully left; the sound comes from the left speaker. Many people believe reflections off the car's sides can make the stage wider than the car. He says they can't. So in practice: once the delays are set and left and right are matched, turn the balance fully left, then fully right. That shows the true edges of your stage.

5.7 #5 The first reference a tuner needs is two speakers in phase, not a concert.

> Many tuners have never sat between two speakers and flipped one's polarity; buy any $5 matched pair at a thrift store, sit centred in an equilateral triangle and hear that imaging is just what two similar speakers do. RARE
>
> **Sources:** B1 @41:56 · B1 @46:15

In plain English: Many people believe a tuner's most important reference is knowing what live music sounds like, so they should go to concerts. He agrees a reference helps, but says the first thing to learn is simpler: what two speakers sound like in phase versus out of phase. "In phase" means both speakers move in and out together. "Polarity" is which way round a speaker is wired; flipping one makes it move opposite to the other, and many tuners have never heard that difference. His exercise: buy any working, matching pair of speakers at a thrift store for about $5. They "do not have to be great", only the same. Set them up so the two speakers and your seat form a triangle with equal sides, sit exactly in the middle and play music. You will hear sound appear between the speakers. His point is that imaging isn't magic; it is simply what two similar speakers do for a listener in the centre.

5.7 #6 How experienced listeners judge a car in seconds, and why multichannel differs.

> Full, impactful bass from the front; a centred singer; width and depth (partly the recording's). Two-channel sounds like a "diorama on the dash"; multichannel sounds like a space bigger than the car, for two people at once.
>
> **Sources:** B1 @53:21 · B1 @55:28

In plain English: He lists what experienced listeners check first when they sit in a demo car. Is the bass full and impactful, and does it seem to come from the front? Is the singer in the centre? Does the stage have width and depth? He notes that width and depth are partly properties of the recording itself. "Two-channel" means ordinary stereo: just a left and a right signal. "Multichannel" means more channels, for example extra speakers at the sides or rear fed by a processor. A

diorama is a small model scene in a box. His point is that even perfect two-channel sound is like a little scene laid out on the dashboard. Multichannel can sound like a space bigger than the car, and it can do that for two people at once, not only the driver.

5.7 #7 A correctly tuned car makes poor recordings sound better, not worse.

> He rejects the idea that accuracy "exposes" bad recordings; that belief is why ear-only tuning goes wrong. RARE
>
> **Source:** DIYMA 2016-01-28

In plain English: Many people believe a very accurate system is unforgiving: it "exposes" every flaw, so poor recordings sound worse on it. He rejects that. Once a car is tuned correctly, he says, recordings he once thought were terrible on lesser systems often sound more interesting and detailed. "Ear-only tuning" means adjusting a system purely by listening, without measuring. His point is that the "accuracy exposes bad recordings" belief is exactly why tuning by ear alone goes wrong. He doubts that someone tuning by ear in their backyard is really exposing faults in recordings made by professionals.

5.7 #8 Verify with tracks whose placement you know.

> Favourite music can mislead (a vocal may never have been mixed centre); Audiofrog's CD uses band-limited mono noise, instruments at five positions, a voice in two rooms and EDM to check that impact comes from the front.
>
> **Sources:** UMI-1 video @27:01 · OFCAP 19 @2:05:01

In plain English: When you check a tune, you need to know where each sound is supposed to be. His warning is that your favourite song may not tell you. Its singer may never have been mixed in the centre, and you could end up tuning the system to put them there. That is why Audiofrog (his company) made a test CD. "Mono" means the same signal in both channels, so it should appear dead centre. "Band-limited" noise is a hiss containing only one range of frequencies, so you can check, for example, that just the bass sounds centred on the dash. A bass guitar and a piano are each placed at five positions from left to right. One track has a person talking and singing in two completely different rooms. EDM (electronic dance music) tracks check that the punch of the bass comes from the front. He says the tracks let installers without a deep musical reference confirm that left is left and the centre is the centre.

5.7 #9 His reference tracks.

> Van Morrison "Lonely Avenue" and "Ancient Highway", Herbie Hancock "Sly" and "Rockit", Marcus Miller "Blast" for midbass, Lyle Lovett "Flyin' Shoes" for lossy tests; judge with older music you know well. Without a centre image he can't even judge tone.
>
> **Sources:** SQOLOGY Pt 2 @46:07 · OFCAP 19 @3:43:08

In plain English: These are songs he uses to judge a system. He chose "Lonely Avenue" for its excellent recording, great imaging, very high crest factor (a big gap between the loudest peaks and the average level) and his emotional connection to the song. "Midbass" is the upper part of the bass range. "Lossy" formats, such as MP3, make files smaller by throwing away sound the format's designers judged hard to hear; he uses "Flyin' Shoes" for its high-frequency ambience when

comparing them with the original. His advice is to judge with older music you know well. The last sentence matters most. He will forgive flaws in tone, such as a peak or dip in the frequency response, but without a centre image he can't even judge whether the tone is right.

5.7 #10 Some listeners genuinely dislike a centre image.

> Of two famous producers, one loved a centred demo car and the other heard it as "mono"; an earbud listener hated a pro SQ tune. Save the correct tune as a preset and, to blur the image, shrink delays rather than deleting them.
>
> **Sources:** B2 @41:19 · OFCAP 19 @1:14:04 · OFCAP 38 @1:02:06

In plain English: A centre image is not everyone's taste. He demoed a car with a clear centre image to two very famous record producers. One loved it. The other heard the centred sound as "mono", as if it came from one point, and hated it. Another listener, who did most of his listening on earbuds, hated a professional SQ (sound quality) tune. "Delay" is a processor setting that holds back the sound from the nearer speakers so that sound from every speaker reaches your seat together; it is what puts the singer in the centre for one seat. His advice: save the correct tune as a preset (a stored setting), because listeners often grow into it later. If the customer wants a less focused image, make the delays smaller rather than deleting them. He explains that delays also keep the bass coherent, and "nobody hates coherent bass".

5.7 #11 Genre and mastering change what drivers receive.

> EDM sends less power to mids and tweeters than pink noise, jazz more; compression belongs in the player, not the master, and the loudness war raises average power on speakers.
>
> **Sources:** Audiofrog blog Dec 2020 · OFCAP 19 @23:02

In plain English: Pink noise is a test hiss with equal energy in every octave (every doubling of frequency); it is commonly used to test speakers. His point is that real music doesn't spread its power the same way. EDM (electronic dance music) sends less power to the midrange speakers and tweeters (the small speakers for high notes) than pink noise does; jazz sends more. "Mastering" is the final step of making a record, where its overall sound and loudness are set. "Compression" squeezes the loud and quiet parts closer together, so the average level rises. The "loudness war" is the long trend of making records ever louder this way. He says it raises the average power the speakers receive. He says compression is useful, but argues it belongs in the playback device, not baked into the recording.

5.7 #12 Distortion on only some tracks is usually in the source.

> Many streaming mixes are clipped and phone "volume levelers" add more; if it persists with the phone turned well down, it's the track or the codec. Often the fix is simply turning the phone down.
>
> RARE
>
> **Sources:** DIYMA 2021-10-11 · DIYMA 2021-07-08

In plain English: Distortion is a harsh, gritty sound that wasn't in the original music. "Clipping" happens when a signal is pushed past the maximum level a device can handle, so the tops of the sound wave are chopped off, which sounds harsh. Many people hear distortion and assume a speaker or amplifier is at fault. He says that when it happens on some tracks and not others, the cause is usually the source. He suspects many mixes made for streaming are already clipped, and

that phone features called "volume levelers" add more. His test: turn the phone's volume well down. If the distortion is still there, it is in the track itself or in the codec (the method used to compress music for streaming over mobile data or Bluetooth). Often, he says, the fix is simply turning the phone down.

5.7 #13 Know when to stop: "imagine how much better..." is the biggest threat to satisfaction.

> Keep a notepad, log tracks that annoy you, check that frequency with pink noise, and fix only real peaks; cutting everything that sometimes annoys you ends in a "smile" curve that sounds bad. RARE
>
> **Sources:** DIYMA 2022-03-03 · DIYMA 2021-08-30

In plain English: Tuning can go on forever, so this is about when to stop. He says the biggest threat to being satisfied is the thought "imagine how much better it would sound if...", along with forum claims that some new trick made all the difference. Recordings differ, and he advises against retuning the car for every one. His method: after tuning, keep a notepad in the car. When something annoys you, write down the track and the time. Later, play that passage while watching an RTA (real-time analyzer, a screen showing how loud each frequency is) to find the offending frequency. Then play pink noise (a test hiss) and check your system's curve there. If there is a real peak (a frequency that is too loud), fix it; if not, the sound is in the recording. A "smile" curve is an EQ shape with the middle turned down, so on a graph it looks like a smile. Turning down everything that sometimes annoys you ends there, and it sounds bad.

5.7 #14 Hearing fades with age, so pair experience with a microphone.

> Most men his age hear little above about 15 kHz (he hears to about 16 kHz).
>
> **Sources:** OFCAP 19 @1:35:05 · DIYMA 2016-02-01

In plain English: Hz (hertz) measures pitch: low numbers are deep bass, high numbers are treble. 15 kHz is 15,000 Hz, very high treble. The ability to hear the highest frequencies fades with age. He says most men his age don't hear much above about 15 kHz, and he himself hears to about 16 kHz. His point, made while agreeing with another forum member, is that even "golden ears" fade, so experience is best combined with a measurement microphone rather than relying on hearing alone.

5.7 #15 How we localize, and why "height EQ" fails.

> Arrival-time cues (about 80–800 Hz), a weak zone to about 1.6 kHz, then level cues; little localizes below 80 Hz. HRTF is about moving heads and moving sounds, so EQ peaks to "raise the stage" misunderstand it. RARE
>
> **Sources:** DIYMA 2016-04-27 · DIYMA 2020-10-14

In plain English: "Localizing" means working out where a sound comes from. He explains that our two ears use different clues in different frequency ranges. From about 80 to 800 Hz, the brain compares when the sound arrives at each ear. From about 800 Hz to 1.6 kHz (1,600 Hz) is a weak transition zone where localizing is poor. Above that, the brain mainly compares how loud the sound is at each ear. Below about 80 Hz, we can hardly tell direction at all. HRTF (head-related transfer function) describes how your head and ears change a sound depending on where it comes from. Some tuners add EQ peaks and dips that copy these height clues to "raise the stage". He says

this misunderstands HRTF, which is about moving heads and moving sounds. A fixed EQ change works only while you are making it; afterwards your brain accepts it as normal tone, and the stage settles back down.

5.7 #16 Your head never stops moving, which hides treble comb filters.

> Above about 1 kHz narrow peaks and dips shift with an inch of movement; that's why sine tones mislead and band-limited pink noise or music doesn't. Car reflections arrive too early to be echoes; they become image spread and tone.
>
> **Sources:** Audiofrog blog Jan 2021 · DIYMA 2017-07-12 · DIYMA 2015-11-01

In plain English: When a sound reaches you both directly and as a reflection off a nearby surface, the two copies add up at some frequencies and cancel at others. On a graph this makes a row of narrow peaks and dips like the teeth of a comb, so it is called a comb filter. In the treble, above about 1 kHz (1,000 Hz), these peaks and dips shift when your head moves even an inch. Because your head is always moving slightly, you don't hear them with music. A sine tone is a single pure frequency, so it can sit right in one peak or dip and make it obvious. That is why he says sine tones mislead, while music, or pink noise limited to one band, spreads across many frequencies and doesn't. In a car, reflections arrive too soon after the direct sound to be heard as echoes. Instead, he says, they spread the image and change the tone.

5.7 #17 Immersive multichannel wins ordinary listeners, and nobody is really a "purist".

> Upmixing is for music too (use 5.1, not 3.1); real homes run multichannel and so does his JBL home system 95% of the time; an L-Acoustics multichannel concert was "a comforter of sound". Anyone who uses a bass control isn't a purist. RARE
>
> **Sources:** OFCAP 38 @41:02 · DIYMA 2021-09-09 · B2 @48:18

In plain English: "Immersive" sound surrounds you instead of coming only from the front. An upmixer is a processor that takes an ordinary two-channel stereo recording and spreads it across more speakers, such as a centre speaker and side or rear speakers. "5.1" means five main channels (front left, centre, front right and two surrounds) plus a subwoofer; "3.1" has only the three front channels plus a subwoofer. He says upmixing isn't just for movies; it works for music too, and he says to use 5.1, not 3.1. Non-audiophiles who heard his upmixed demo car said the sound was everywhere, yet the singer and instruments were still in front of them. He notes that real homes run multichannel, and his own JBL home system plays music in multichannel 95% of the time. A multichannel concert on an L-Acoustics system felt to him like "a comforter of sound". Many people believe a "purist" must play stereo only as stereo. His reply: anyone who uses a bass control isn't a purist either.

5.7 #18 A binaural dummy head is a listening and teaching tool, not a tuning shortcut.

> Its response changes with angle, it conveys imaging but not loudness or impact, and tuning with an "ears" mic would need a whole new target.
>
> **Sources:** CMA 2023 @55:01 · DIYMA 2021-11-23

In plain English: A binaural dummy head is a model of a human head with a microphone in each ear. Recordings made with it, played on headphones, sound much like being there. He sees it as a listening and teaching tool. For example, a shop can record a finished car with it so people can

hear the result on headphones. Those recordings show imaging, but they can't convey how loud the system is or how hard the bass hits. It is not a shortcut for tuning. Its measured response changes with the angle the sound arrives from, so each speaker location would need its own correction. A "target" is the ideal frequency-response curve you tune toward. Tuning with an "ears" microphone would need a whole new target, and he says there is nothing to gain from it.

5.7 #19 Don't measure the top octave through Bluetooth, MP3 or a laptop's own output.

> Most "nothing above 10 kHz" complaints come from 128k rips or Bluetooth; laptops can filter test signals in the OS and phones can clip at full volume, so use an external sound card.
>
> **Sources:** testgear.audiofrog.com (COMPANY) · DIYMA 2021-12-05 · DIYMA 2021-12-22

In plain English: An octave is a doubling of frequency; the "top octave" is the highest treble, roughly 10 to 20 kHz. When people measure their car with test noise, a common complaint is that the microphone shows nothing above 10 kHz. He says most of these cases come from the test signal, not the speakers: the CD was ripped to a low-quality 128k MP3, or the signal was sent over Bluetooth. Both are "lossy" (they throw away some high-frequency sound to save data). Laptops can also filter test signals inside the operating system, as speaker protection that can't be switched off, and phones can clip (distort) at the top of their volume range. So in practice: play test signals through an external sound card.

5.7 #20 The best "hi-fi" sound he ever heard came from a pro line array.

> Ray LaMontagne on JBL VerTec beat anything at a hi-fi show; loud and clean aren't opposites.
>
> **Source:** OFCAP 19 @3:45:01

In plain English: A line array is a tall column of speakers hung at a concert venue to cover a large audience. JBL VerTec is a professional line-array system. He heard singer Ray LaMontagne at Town Hall in Manhattan through one and found it more "hi-fi" than anything at a hi-fi show, where expensive home systems are demonstrated. His point is that sound can be loud and clean at the same time; the two aren't opposites.

5.8 Who Andy is (profile)

What this part is about: A short profile of Andy Wehmeyer's career, so you can judge where his advice comes from.

- **Installer first.** Built speakers as a teen, was a "terrible" installer at an El Paso shop, became its box builder (box software on a DOS PC around 1989–90) and spent about ten years at retail, ranking in Installation News's top 25–50. SQOLOGY Pt 1 @6:03 · B1 @1:57 · B2 @7:27
- **About 18 years at Harman from about 1995–96.** Hired by Mike Giffin as Infinity class-D trainer on his former boss's recommendation; trained as far as the Audax factory in France; worked beside Richard Small, Floyd Toole and Sean Olive. SQOLOGY Pt 1 @11:00 · B1 @11:47
- **Product leader.** Global Product Line Manager for JBL/Infinity/Harman Kardon car audio; drove MS-2 and MS-8 (the MS-8 auto-EQ was designed by Ulrich Horbach, he notes); designed the Infinity Perfect 10 VQ sub and an analog all-pass imaging switch in the Basslink. SEMA 2012 · DIYMA 2017-06-28 · DIYMA 2021-12-15

- **DSP user since 1992 and co-inventor.** Co-holder of US patent 9,241,230 (extracting channels from factory systems) and an auto-EQ application; "not a coder or PhD acoustician" but qualified to teach tuning. DIYMA 2020-10-04
- **Left Harman in March 2013** as priorities moved to cheaper products for emerging markets. DIYMA 2013-03-02 · SQOLOGY Pt 2 @26:05
- **Co-founded Audiofrog** with Gary Biggs ("GB") and Marcus Yeo; dealer-direct; calls himself its "customer service guy". **Conflict:** founding year given as 2011, 2014 or 2012 in different sources; products shipped by mid-2015 and a 10th anniversary was marked in May 2023. BusinessWorld · OFCAP 38 · DIYMA 2021-12-18
- **Origin of the "Harman" curve:** his retail-installer curve became the MS-8 target and was later confirmed by Harman research. DIYMA 2020-10-14 · Audiofrog blog Mar 2021
- **A teacher by vocation:** SEMA "Car Audio Mythbusters" (2012), KnowledgeFest tuning classes, Mobile Electronics columns, Audiofrog U, the Tech Blog and thousands of forum posts; in 2023 he shifted from teaching tuning to teaching system design (mapping a car with a mic and speaker before choosing locations). CE Outlook 2015 · B1 @1:00:14
- **Modest competition record by choice:** tuned Team JBL cars with Gary Biggs; multichannel Mini and Volvo entries won shows. DIYMA 2011-04-16 · DIYMA 2021-09-09
- **Self-description:** an "armchair acoustician" and "integrator, not a creator" who still finds phase plots confusing after 35 years. DIYMA 2011-06-09 · B1 @1:07:56
- **Caveat:** he sells much of what he recommends (speakers, amps, test gear, an upmixer DSP); weigh product-specific claims accordingly, though he steers buyers elsewhere when Audiofrog doesn't fit. OFCAP 19 @2:51:01

In plain English: Andy Wehmeyer learned car audio hands-on. He built speakers as a teenager, then spent about ten years in car-audio retail, starting as an installer at an El Paso shop and becoming its box builder. He then spent about 18 years at Harman, the company behind JBL, Infinity and Harman Kardon car audio, where he managed those product lines, drove the MS-2 and MS-8 processors and worked beside researchers such as Floyd Toole and Sean Olive. He left Harman in March 2013. He co-founded Audiofrog (sources differ on the year). His background matters because his advice comes from both the installer's bench and the engineering lab, and he has taught it in classes, columns and thousands of forum posts. He calls himself an "armchair acoustician", not a PhD acoustician. He also sells much of what he recommends, so weigh product-specific claims accordingly, though he steers buyers elsewhere when Audiofrog doesn't fit.

5.9 Top rarely-discussed points in this section

1. Too little power can't blow a speaker; clipping roughly doubles average power and dumps it on the tweeters (5.1 #26).
2. With music a tweeter needs a watt or two; a big amp at full output can cook a passive tweeter without clipping (5.1 #28).
3. Amps with built-in DSP can leave a "1,000 W" amp making about 10 W after tuning; use separate DSP and amps for high-end work (5.2 #8).
4. About 70% of GB25 (2.5" midrange) warranty claims are DSP input clipping; set the DSP input so it can never clip (5.4 #12, 5.1 #61).

5. Move the image with level, never delay; 6 dB moves the centre halfway (5.1 #62).
6. "Bass up front" is crossover phase: lower the LR4 sub low-pass half an octave per 12 dB of sub boost (5.1 #65).
7. Only about 10–20% of a factory EQ corrects the speakers, so upgrade speakers on the factory EQ first; never sum a Logic 7-type system back to flat (5.3 #6, #7).
8. Factory systems are PhD-engineered; aftermarket people judge them on bass and miss their two-seat imaging (5.5 #16).
9. Buy from the best shop in town but ask it for a mid-level job (5.4 #1).
10. "Correct" is only the start of expertise; the disco customer shows when to back off it (5.4 #15).
11. Learn imaging with a $5 thrift-store pair before chasing concert references (5.7 #5).
12. Separate L/R amps and channel-separation specs are irrelevant next to in-car crosstalk (5.1 #24).
13. Break-in matters only for subs (5.1 #30).
14. "Individually matched" pairs signal a high-variance process (5.1 #35).
15. Premium buyers are 35–70 and decide with a partner, which shaped the Audiofrog name (5.5 #14).
16. Upmixers are missing from DSPs because of product managers, not licence cost (5.5 #20).
17. A correctly tuned car makes bad recordings sound better (5.7 #7); know when to stop tweaking (5.7 #13).
18. Distortion on only some tracks is usually clipped streaming masters or the phone (5.7 #12).

Changed views and conflicts in this section

- **Changed view, rear fill:** 2017 forum recipes for differential (L−R) rear fill; by 2021 he advises plain stereo rears and says he may have helped make L−R fill semi-mythical (5.1 #75).
- **Shift of emphasis, efficiency vs power:** "power is cheap" (2008–2013, arguing for active crossovers) vs efficiency first for very loud systems (2021); different contexts (5.2 #10).
- **Shift of emphasis, speakers' importance:** 2008–2011 "speakers matter little if not badly designed"; 2016 onward, no DSP can remove a bad speaker's distortion (5.3 #3).
- **Shift of teaching focus:** from tuning (about 20 years) to system design and measurement literacy in 2023 (5.8).
- **Conflict, gain-setting tracks:** 2015 endorsement of −10 dB/multitone gain setting vs 2021 warning against using such tracks to max treble gains (5.1 #27).
- **Conflict, CES 1996 blind test numbers:** three listeners after 700–800 demos (2022) vs four after about 1,500 (2023) (5.1 #21).
- **Conflict, LR4 vs 48 dB/oct protection:** his claim that LR4 protects better is argued only from the "knee"; a note-taker could not reconcile it with attenuation one octave down (5.1 #57).
- **Conflict, Audiofrog founding year:** 2011, 2012 or 2014 depending on source (5.8).
- **Conflict, tuning-guide structure:** one description has the process in the first two pages followed by about 60 pages of explanation (2018 forum); another has about 40 pages of explanation followed by about 20 of process (Appendix A).

- **Consistent views (no change found):** damping factor (2008–2021), break-in (refined, never reversed, 2015–2021; see 2.8 #6), tape-measure delays (2015–2021), cone-material scepticism (2013–2024).

Glossary

Plain-English definitions of terms used in this guide. The definitions are general background written by the compiler. Where Andy Wehmeyer uses a term in a particular way, a note marked How Andy uses it says so, with a reference to the point where you can read his full view and its sources.

A-pillar: the post on each side of the windshield. Speakers are often mounted in pods there to raise the sound. *How Andy uses it:* he calls it "a HORRIBLE place for a midrange", though his company's own sales copy promotes an A-pillar system (1.6 #6).

A/D converter (analog-to-digital converter): the part of a DSP that turns the incoming electrical music signal into numbers. If the incoming signal is too strong, it clips. *How Andy uses it:* he says the DSP's input-sensitivity control has "ONE function": keeping the A/D converter from clipping at maximum volume (4.8 #1).

Acoustic crossover: the point where two speakers actually hand over in the air, as measured with a microphone. It depends on both the crossover filter and the speakers' own responses, so it is often not at the frequency set on the DSP. *How Andy uses it:* turning the sub up moves the acoustic crossover up and out of phase, which is why he lowers the sub's low-pass as he raises its level (4.2 #6).

Acoustic reciprocity: the principle that a sound path works the same in both directions, so a speaker and a microphone can swap places. *How Andy uses it:* he "maps" a car by putting a speaker at the driver's head and a microphone at each possible speaker location; this works for sub, midbass and midrange spots, not tweeters (2.3 #13).

Active crossover (active system): a crossover that works on the signal before the amplifiers, usually inside a DSP, with a separate amplifier channel for each speaker. Every setting can be adjusted.

All-pass filter: a filter that shifts the timing (phase) of sound in one frequency band without changing its loudness. *How Andy uses it:* he describes an all-pass near 250 Hz as a way to improve the image in both front seats at once, where delay helps one seat and hurts the other (1.9 #4, 1.9 #7). He also says some factory systems use all-pass filters in place of delay, and that you should not add delays on top of them (4.14 #2, 4.14 #4).

Amplifier (amp): the unit that makes the music signal strong enough to drive speakers. Its output is rated in watts.

Baffle: the flat panel a speaker is mounted on, which keeps the sound from the front of the cone apart from the sound off its back. *How Andy uses it:* he says a door midbass needs a sealed baffle, not a box (2.1 #1).

Balance control: the control that shifts sound between the left and right speakers. *How Andy uses it:* panning it fully left, then fully right, shows the real edges of the stage (1.7 #1).

Bandwidth: the range of frequencies a speaker, filter or system covers.

Biquad: a standard building block of digital filters; one biquad can make one EQ band or one section of a crossover.

Boundary microphone: a microphone mounted flush with a surface, so it does not pick up a reflection from that surface. Andy describes using a 3D-printed one for mapping speaker locations (2.3 #13).

Break-in: the idea that new speakers change as they are played. *How Andy uses it:* he says the real change is resin in the spider cracking, which matters mainly for subwoofers; mids and tweeters crossed above resonance barely change (5.1 #30).

Cabin gain: the boost a closed car gives to deep bass, rising as frequency falls. *How Andy uses it*: he calls it the car's "transfer function" (in-car response divided by the woofer's outdoor response) and says it belongs to the car, not the woofer; build the most efficient box you can fit, then EQ (4.1 #1, 4.1 #6).

Center channel (center speaker): a speaker in the middle of the dash meant to anchor centered sounds such as vocals. *How Andy uses it:* "L+R is not a center"; a real center plays only what left and right share, extracted by an upmixer (1.10 #1, 1.10 #2).

Channel separation: how little of one channel leaks into the other inside a piece of equipment, in dB. *How Andy uses it:* he says it makes no difference to imaging in a car: at 60 dB of separation, 1 W leaks only a millionth of a watt into the other channel, while each ear already hears the opposite speaker at maybe −6 dB (5.1 #24).

Class-D amplifier: an efficient type of amplifier that switches its output on and off very rapidly; most modern car amps are class-D.

Clipping: what happens when any part of the system is asked for more than it can give: the tops of the sound waves are cut off flat. *How Andy uses it:* he says clipping, not too little power, is what destroys tweeters, and that DSP and head-unit clipping is the usual culprit (5.1 #26, 4.7 #6).

Coax: short for coaxial cable, a signal cable with one central conductor inside a surrounding shield; also short for a coaxial speaker, with a tweeter mounted in the middle of a woofer. *How Andy uses it:* he says coax cable suits single-ended amp inputs, and twisted pair suits differential ones (4.10 #5).

Codec: the method used to compress and decompress digital audio for streaming or storage.

Comb filter: a pattern of closely spaced peaks and dips caused when the same sound arrives at two slightly different times (from two speakers, or a speaker and its reflection). On a graph it looks like the teeth of a comb. *How Andy uses it:* he says "width" from delay tricks is really comb filtering, and that no EQ or delay can fix a reflection's comb (1.7 #2, 1.3 #4).

Cone: the moving part of a woofer or midrange that pushes the air.

Correlated and uncorrelated noise: test noise that is identical in both channels (correlated, also called mono) or different in each channel (uncorrelated). *How Andy uses it:* he reads phase from mono (correlated) pink noise; his later view is that uncorrelated noise is useless except for upmixers, L−R rears or gaming an RTA (3.8 #7, 3.3 #4).

Cross-car mode: a bass cancellation caused by sound reflecting off the far side of the car. *How Andy uses it:* he links a 60–80 Hz dip in one midbass to it and says to cross above it to the sub rather than EQ it; he describes every car having one near 70 Hz (4.6 #1).

Crossfiring: aiming each speaker across the car toward the opposite seat. *How Andy uses it:* he calls crossfired pillar tweeters a "band-aid" (1.5 #6).

Crossover: a filter that splits the music between speakers so each plays only the range it handles well. The "crossover frequency" is where one speaker hands over to the next.

D/A converter (DAC): the part of a DSP that turns numbers back into an electrical signal for the amplifiers. Its output cannot go above 0 dBFS.

Damping factor: a specification describing how firmly an amplifier controls a speaker's motion. *How Andy uses it*: he says anything above about 10 is enough; a damping factor of 1000 moves Qts only from 0.707 to 0.708 (5.1 #19).

Dash: the dashboard. Many factory systems have speakers in the dash top.

DATS: a commercial test kit for measuring a speaker's impedance and Thiele/Small parameters.

dB (decibel): the unit for differences in level. About 3 dB is clearly noticeable, 6 dB is a big change, and about 10 dB sounds roughly twice as loud. Each 3 dB more output needs about twice the amplifier power.

dB/oct (decibels per octave): the steepness of a filter: how quickly it turns sound down outside its range. A 24 dB/oct filter turns sound down by 24 dB for each octave beyond its corner.

dBFS (decibels relative to full scale): a level in the digital domain, where 0 dBFS is the largest possible digital signal. Anything that would go above it clips.

Deadening (sound deadener): heavy, sticky damping sheets and foam applied to car panels; Dynamat is one brand. *How Andy uses it*: he uses it to stop rattles and panel movement, done in order: the door trim panel first, then the inner skin (2.2 #1).

Delay (time alignment): holding back one speaker's sound by a few thousandths of a second so it arrives at the listener with the others. *How Andy uses it*: "Delay doesn't move speakers"; he sets delay from tape-measured distances and centers the image with level, not delay (3.9 #1, 3.9 #3, 3.10 #2).

Differential input: an amplifier input that measures the difference between two signal wires, which helps reject noise. *How Andy uses it*: he identifies it by measuring hundreds of ohms between the RCA shield and the amp's power ground, and pairs it with twisted-pair cable (4.10 #5).

Directivity: how a speaker's sound output changes with direction: how much it "beams" forward instead of spreading out.

Dispersion: how widely a speaker spreads its sound. *How Andy uses it*: his rule compares the driver's circumference with the wavelength; below 1 the sound is the same on- and off-axis (1.4 #2).

Distortion: anything added to the sound that was not in the original signal, such as harshness from clipping or buzzes from rattles. *How Andy uses it*: he says distortion and rattles, not arrival time, give away where bass is coming from (4.2 #2).

DIYMA: DIYMobileAudio, an online car-audio forum where many of his posts appear, under accounts the forum renamed "14642" and "156546".

Driver: a single speaker unit (tweeter, midrange, woofer or subwoofer).

DSP (digital signal processor): a small computer for sound that provides EQ, crossovers, delay and level control for each speaker channel.

DSP amplifier: an amplifier with a DSP built in. *How Andy uses it*: he says it has no analog gain after its D/A converter, so output lost to EQ cuts cannot be won back (4.8 #9).

Dust cap: the dome in the middle of a woofer's cone. Andy measures delay distances to the center of each dust cap or grille (3.9 #3).

Enclosure: a speaker box. *How Andy uses it*: he says a pod is an enclosure too, and must be treated as one (2.4 #1).

EQ (equalization): turning particular frequency bands up or down to correct a system's tonal balance.

Excursion: how far a speaker's cone moves in and out. More excursion means more strain and, past a point, more distortion.

F3: the frequency where a speaker's (or box's) bass output has fallen 3 dB. *How Andy uses it*: he says matching a sub's F3 to the car is "a ridiculous crapshoot" (4.1 #6).

Factory system (OEM system): the audio system a car is built with; OEM means "original equipment manufacturer". Many include their own processing.

Far-side bias: the sound image being pulled toward the speaker on the far side of the car. *How Andy uses it:* he names it as a result of the off-axis "hole" a big woofer and a small tweeter leave at their crossover in door locations, and says a 3-way reduces it (1.4 #4).

Fc: a speaker's resonance once it is mounted in a sealed box (higher than its Fs).

FFT: Fast Fourier Transform, the math that turns a recorded signal into a frequency graph. Measurement software lets you choose its size (for example 16384 or 32k).

FIR filter: "finite impulse response", a type of digital filter that can change timing and level very precisely but adds processing delay (latency).

Frequency response: a graph or description of how loud a speaker or system plays at each frequency.

Fs: a speaker's free-air resonance, the frequency at which its cone naturally moves most easily. Below it, the speaker works less well and distorts more.

Gain: an amplifier's input control. It matches the amp to the strength of the incoming signal; it does not set power. *How Andy uses it*: his gain procedure uses a sine wave on the sub amp only and sets the rest by pink noise and RTA (4.8 #4).

Gasket: a soft seal between a speaker and the surface it mounts to, to stop air leaks and rattles.

Ground loop: noise caused by equipment sharing ground connections at different voltages. *How Andy uses it*: ground loops cause whine, not hiss (4.10 #1).

Group delay: how much a filter or speaker delays different frequencies. *How Andy uses it*: he calls group delay "less a curse than people think" and says it is overrated in cars.

Haas effect (precedence effect): when the same sound arrives twice close together, we hear a single sound located at the first arrival. *How Andy uses it*: he says car reflections arrive too early to be heard as echoes, so instead they spread the image and change the tone (5.7 #16).

"Harman curve": a popular target curve for car audio tuning, named after the company Harman. *How Andy uses it*: he says it began as his own installer curve, which he brought to Harman, where it became the MS-8 curve (3.5 #2).

Harmonics: extra tones at whole-number multiples of a note's frequency (twice, three times…). Distortion adds harmonics. *How Andy uses it*: his test adds 2nd and 3rd harmonics to a bass track to show that distortion makes bass easy to locate (4.2 #2).

Head unit: the radio or main audio unit in the dash, also called the source.

High-pass filter: a filter that lets frequencies above a chosen point through and turns lower ones down. It keeps deep bass out of small speakers.

Hiss and whine: hiss is a steady "shhh" noise; whine is a pitched noise that often rises and falls with engine speed. *How Andy uses it:* "Hiss is not a ground loop; loops whine" (4.10 #1).

Hz and kHz: hertz, cycles per second, the unit of frequency (pitch). A kilohertz (kHz) is 1,000 Hz. Hearing spans roughly 20 Hz to 20 kHz.

Imaging: the illusion that each voice or instrument sits at a precise spot in front of you. *How Andy uses it*: he suggests learning what it is with any matched cheap pair of speakers (5.7 #5).

Impedance: a speaker's electrical resistance to the amplifier's signal, measured in ohms; it varies with frequency.

Impedance sweep: a measurement of impedance across frequency. *How Andy uses it*: he uses one to check a pod, where a leak shows up as an extra, port-like resonance (2.4 #4).

Impulse response: a measurement of how a system responds over time to a very short click. *How Andy uses it:* he says its peaks are not true arrival times for subs and midbass, so a tape measure is usually better (3.9 #5).

Inductance: the electrical property of a speaker's voice coil that resists fast changes in current; it affects how a speaker behaves at higher frequencies.

Infinite baffle (IB): mounting a speaker on a large panel with no sealed box behind it, so a large space (such as a trunk or door) acts as its box. "IB-type" drivers are designed for this.

Inverted triangle: *How Andy uses it*: his picture of stage width, with you at the bottom point and the speakers at the two upper corners; the stage can't be wider than the speakers (1.7 #1).

Kick panel: the panel at the side of the front footwell, a common place for midbass or midrange speakers.

L+R and L−R: the sum of the left and right channels (what they share, mono) and the difference between them (what differs). *How Andy uses it*: L+R alone is not a center (1.10 #1); his later view is that L−R rear fill sounds "phasy" (1.11 #5).

Latency: the delay processing adds between the input and the output. Long FIR filters add enough to affect hands-free calls.

Level: how loud a channel or speaker is set relative to the others. *How Andy uses it*: he centers an off-center image with level, never delay (3.10 #2).

LOC (line output converter): a device that turns a speaker-level signal (for example from a factory radio) into a lower, RCA-level signal for an amplifier or DSP.

Logic 7: Harman's processing that spreads stereo across many speakers, used in some factory systems. *How Andy uses it*: "DO NOT TRY TO SUM" a Logic 7-type car back together (4.15 #4).

Low shelf: an EQ filter that raises or lowers everything below a chosen frequency by a set amount. *How Andy uses it*: he builds his target's bass lift as a 12 dB/octave low shelf at about 60 Hz (3.5 #1).

Low-pass filter: a filter that lets frequencies below a chosen point through and turns higher ones down. Subwoofers use one.

LR4 (Linkwitz-Riley, 4th order): a common crossover filter type with a 24 dB per octave slope. *How Andy uses it*: his default sub/midbass crossover is 60 Hz on the sub and 80 Hz on the midbass, both LR4 (4.2 #6).

Matrix: a simple circuit that mixes left and right into sum (L+R) and difference (L−R) signals. *How Andy uses it*: he says passive L+R/L−R matrices "do not work" for a center (1.10 #1), and he later judged matrix upmixers poor for music (1.12 #6).

Microfarad (µF): the unit used for capacitor size. A series capacitor blocks low frequencies; a smaller value blocks more.

Midbass: the upper-bass and lower-middle range, roughly 80–300 Hz, and the speaker (usually a 6–8 inch woofer in the door) that plays it.

Middler: *How Andy uses it*: his name for a factory dash speaker aimed at the windshield, which carmakers use as a "poor man's tweeter"; with raked glass its highs reflect to every seat, but it fails with upright glass, as in a Jeep (1.6 #9, 2.3 #9). His praise of it sits beside his criticism of firing mids up at the glass; the conflicts list in "Start here" sets out both (1.6 #23).

Midrange: the middle frequencies, where most of the human voice sits, and the small speaker (about 2–4 inches) that plays them.

Minimum phase: a type of response in which level and timing are tied together, so correcting the level with EQ also corrects the timing. *How Andy uses it*: box and driver peaks are minimum phase and respond to EQ; reflection dips do not (3.6 #10).

Mono: the same signal in both channels.

MS-8: the JBL MS-8, a car-audio processor with automatic multi-microphone tuning, made by Harman. Andy was its product manager.

Near-field: listening or measuring close to a speaker, where its direct sound dominates the room's reflections. Studio monitors are near-field speakers.

Null: a deep dip in the response where two sounds cancel. *How Andy uses it*: never significantly boost one with EQ (3.6 #9).

Octave: a doubling of frequency, such as 100 to 200 Hz.

Off-axis and on-axis: on-axis is straight in front of a speaker; off-axis is anywhere to the side, above or below.

Ohm (Ω): the unit of electrical resistance and impedance. Car speakers are commonly 2 or 4 ohms.

Omnidirectional: radiating sound equally in all directions. Speakers are omnidirectional at frequencies whose wavelength is large compared with the driver.

Parametric EQ: an equalizer in which you set each band's center frequency, its boost or cut, and its width (Q).

Passband: the range of frequencies a speaker is meant to play once the crossover has been set.

Passive crossover: a small set of capacitors, coils and resistors between the amplifier and the speakers that splits the sound. Simple, but fixed.

Phase: where a wave is in its cycle, in degrees. Two speakers in phase add up; 180° apart, they cancel. *How Andy uses it:* the goal is "far from 180°", not 0°; anything within ±90° can be left alone (3.8 #6).

Pink noise: a hiss with equal energy in every octave, used as a test signal because its balance is similar to music's.

Pod: a custom housing for a speaker, often on an A-pillar or dash. *How Andy uses it:* a pod is an enclosure and must be sized, sealed and gasketed (2.4 #1, 2.4 #3).

Polarity: which speaker wire is plus and which is minus. Reversing it on one speaker makes it move backwards relative to the others.

Port (vented box): a tube or slot in a speaker box that reinforces bass around a chosen tuning frequency. *How Andy uses it:* he prefers vented sub boxes tuned for minimum excursion, with the in-car peak EQ'd out (4.3 #1).

Power response: a speaker's output averaged over all directions. *How Andy uses it:* because a car has no reverberant field, he says you mostly hear each speaker's power response (1.3 #1).

Q: for a speaker, a number describing how peaky its bass resonance is (low Q is flatter); for an EQ or filter, how narrow its band is (high Q is narrow). *How Andy uses it:* he recommends a flat, low-Q midbass in cars (2.1 #9).

Qts and Qtc: Qts is a speaker's total Q in free air; Qtc is its Q once mounted in a sealed box.

Rattle: a buzz from a loose panel, wire or part vibrating with the music. *How Andy uses it:* he says rattles and distortion give away a sub's location (4.2 #1).

RCA cable: the common signal cable between a source or DSP and an amplifier, named after its plug.

Rear fill: rear speakers played quietly to add space without pulling the sound backwards. *How Andy uses it:* plain in-phase stereo, high-passed, raised, then delayed until the stage returns to the front (1.11 #4).

Reflection: sound bouncing off a surface such as glass or the dash. *How Andy uses it:* he predicts reflection dips with a tape measure (1.3 #5).

Resonance: the frequency at which something (a speaker, a panel, a box, a room) vibrates most easily.

Reverberant field: the late, diffuse sound that builds up in a large room after many reflections. *How Andy uses it:* he says a car has none (1.3 #1).

REW (Room EQ Wizard): free measurement software often used for tuning, usually with a swept tone.

RTA (real-time analyzer): a display that shows the level at each frequency live, usually while pink noise plays. *How Andy uses it:* from 2018 on, his main tool for production tuning is a live RTA on mono pink noise (3.3 #1).

Sail panel: the small triangular panel at the front corner of a front door window, near the mirror; a common tweeter location.

Sealed box: an airtight speaker box with no port. *How Andy uses it:* with little space but plenty of power, sealed plus EQ; with no EQ at all, sealed is probably better (4.3 #4, 4.3 #6).

Sensitivity: how loud a speaker plays from a given power, usually in dB at 1 watt, 1 metre. *How Andy uses it:* a speaker 3 dB more sensitive equals doubling the amplifier (4.7 #3).

Signal-to-noise ratio (SNR): how much louder the music signal is than the background hiss.

Sine wave: a pure single-frequency tone, used for some tests. *How Andy uses it:* in his gain procedure only the sub amp gets a sine wave, and tweeters never do (4.8 #4).

Single-ended input: an amplifier input that measures its signal against ground. *How Andy uses it:* he identifies it by measuring near 0 ohms between the RCA shield and the amp's power ground, and pairs it with coax (4.10 #5).

"Smile" curve: an EQ shape with boosted bass and treble and a scooped-out middle, like a smile. *How Andy uses it:* cutting everything that sometimes annoys you ends in a "smile" curve that sounds bad (5.7 #13).

Smoothing: averaging a measured graph over a band (for example 1/6 octave) so tiny wiggles don't hide the overall shape. *How Andy uses it:* one mic with 1/6-octave smoothing is good enough for production tuning (3.4 #5).

Spatial average: a measurement averaged from several microphone positions around the listening area. *How Andy uses it:* he reports that a smoothed single mic was about 90% as accurate (3.4 #5).

Spider: the corrugated fabric ring that holds a speaker's cone centered and acts as a spring. *How Andy uses it:* break-in is the spider's resin cracking (5.1 #30).

SPL (sound pressure level): how loud a sound actually is in the air, in dB.

SQ (sound quality): car audio aimed at accurate, realistic sound, as opposed to sheer loudness; also the name of competitions judged on it.

Stage (soundstage): the overall picture of where the music seems to be: its width, height and depth. *How Andy uses it:* the stage can't be wider than your speakers (1.7 #1).

Subwoofer (sub): a large speaker in its own box that plays the deepest bass.

Surround: the flexible rubber or foam ring that joins a cone to the speaker's frame. *How Andy uses it:* a surround hitting a grille or panel is an installation error (2.1 #19).

Sweep: a test tone that glides from low to high frequency, analyzed by measurement software.

Tape-measure delay: *How Andy uses it:* setting delays from distances measured with a tape measure from the headrest center (or mic tip) to the center of each speaker's grille or dust cap, keyed to the midrange, then left alone. He converts with ms = inches ÷ 13,584 × 1000 (3.9 #3).

Target curve: the frequency-response shape a tuner aims for. *How Andy uses it:* his target is about +10 dB below about 60 Hz, flat from about 160 Hz, and −6 dB at 20 kHz, placed on the dips with the peaks cut down to it (3.5 #1, 3.6 #3).

Thiele/Small (T/S) parameters: standard numbers describing a speaker's low-frequency behavior, such as Fs, Qts and Vas, used to design boxes.

Top plate: the steel plate at the front of a speaker's magnet structure, through which the voice coil moves.

Transfer function: a measurement of how one thing changes a signal; in car audio, how the car changes a speaker's output. *How Andy uses it:* cabin gain is the car's transfer function (4.1 #1).

Trim panel: the inner plastic or fabric panel of a door. *How Andy uses it:* it is the worst rattle source, so deaden it first (2.2 #1).

Tweeter: a small speaker, usually a dome about 1 inch (25 mm) across, that plays the highest frequencies.

Twisted pair: a cable with two signal wires twisted together, which helps reject noise. *How Andy uses it:* use it for differential inputs (4.10 #5).

Two-way and three-way: a two-way system splits each side between a woofer and a tweeter; a three-way adds a midrange. *How Andy uses it:* he says cars need 3-ways (1.4 #4, 1.4 #6).

Upmixer: a processor that spreads two-channel stereo across more channels, such as a center and rear speakers. *How Andy uses it:* for music he favors upmixers that extract the content left and right share (1.12 #6).

Vas: a speaker's equivalent air volume, a T/S parameter used in box design.

Voice coil: the coil of wire attached to the cone that the amplifier's current flows through; it moves in the magnet's field to drive the cone.

Waveguide: a shallow horn-like shape around a tweeter that controls how it spreads sound. *How Andy uses it:* he says one big enough to match a 6-inch woofer won't fit in a car (1.4 #6).

Wavelength: the physical length of one cycle of a sound wave; low notes have long waves, high notes short ones. *How Andy uses it:* a reflection dip falls where the reflected path is half a wavelength longer than the direct one (1.3 #5).

Widebander (wideband speaker): a small full-range speaker meant to play most of the range without a separate tweeter. *How Andy uses it:* his later view is to aim tweeterless widebanders at the windshield, since on-axis they sound "clinical" (1.8 #4).

Xmax: the distance a speaker cone can travel in one direction while still behaving in a linear (clean) way. *How Andy uses it:* the cone can travel past Xmax, so leave clearance; keep grilles about 1.25 inches off a woofer's front (2.1 #19).

Sources and method

Where the original material lives, and how this research was done.

Appendix A. Where Andy's material lives

Videos, podcasts and livestreams

"Transcribed" = YouTube auto-captions read in full for this research.

#	Title	Channel / host	Date	Length	URL	Best for	Transcribed?
1	Old Fashioned Car Audio Podcast (OFCAP) Ep. 19, "Andy Wehmeyer - AudioFrog"	Matt Schaeffer & Gary Bell	Audio Oct 29, 2021 (YouTube Oct 9, 2025)	3:54:13	YouTube · Apple	Richest single source: power and clipping myths, crossover mistakes, tuning philosophy, choosing a shop, sales, industry economics	Yes
2	OFCAP Ep. 38, "Andy Wehmeyer Phones Home"	Matt Schaeffer & Gary Bell	Audio May 17, 2023 (YouTube Oct 9, 2025)	1:25:44	YouTube	Volvo C30 demo build, mounting and deadening, two-seat philosophy, de-tuning for image-haters, contest gaming	Yes
3	SQOLOGY, "feat. Andy Wehmeyer – Part One"	Benny Z & Company	Audio Nov 27, 2017 (YouTube Sep 25, 2024)	1:03:55	YouTube	Background, MS-8 origin, MP3 and hi-res, experience vs data, OEM vs aftermarket	Yes
4	SQOLOGY, "feat. Andy Wehmeyer – Part Two"	Benny Z & Company	Audio Jan 5, 2018 (YouTube Sep 25, 2024)	48:23	YouTube	Upmixing, two-seat tuning, score sheets, leaving Harman, the Audiofrog name	Yes
5	"AUDIOFROG | AUDIOPHILES | CONNECTED"	CMA Networks (Ben Woo, Importel)	Nov 27, 2023	1:05:07	YouTube	Factory upgrade path, class-D and pink-noise power, immersive audio, binaural demos	Yes
6	"Audiofrog - High Fidelity Car Audio Designed by Experts with Andy Wehmeyer - Importel"	CMA Networks (Ben Woo, Kelly Holly)	Mar 24, 2021	54:45	YouTube	Product-line rationale, durability and power tests, Bose/Harman dash trap	Yes
7	"AUDIOFROG | AUDIOPHILE | SHOWCASE" (A-series amp launch)	CMA Networks	Oct 29–30, 2022	1:15:45	YouTube	Amplifier truths: load-dependent class D, feedback, thermal rollback, FM noise, differential inputs	Yes

#	Title	Channel / host	Date	Length	URL	Best for	Transcribed?
8	"CMA CONNECTED | Audiofrog" (B2)	CMA Networks livestream (Ben Woo, Romesh of Importel, dealer Bentley Burchard)	Oct 25, 2021	1:04:09	YouTube	Premium buyer profile and the name, three kinds of shops, dash "middler" physics, reuse OEM systems, "back off correct", the disco customer	Yes (round 4)
9	"Andy and Audiofrog !" (B1)	Absolute Electronix livestream	Mar 1, 2023 (US evening)	1:13:32	YouTube	Career story, woofers designed from the car, DSP-amp gain trap, class-D filters, $5 imaging lesson, one-to-one OEM upmix integration, 2023 system-design seminars	Yes (round 4)
10	"Andy Wehmeyer is live at a Special time 2:00 pm est"	Dean and Fernando's Car Stereo Clips (same stream also on Five Star Car Stereo: bvd8ugEw7iA)	Jan 9, 2024	1:01:29	YouTube	All-pass filters and the 250 Hz two-seat insight, upmixer history, binaural head	Yes (the Five Star copy was not re-transcribed)
11	"Andy Wehmeyer talks Up mixing" / "...talks All Pass filters"	Dean and Fernando's Car Stereo Clips	Aug 7 / Aug 6, 2024	15:29 / 18:50	Up mixing · All Pass	Excerpts of #10 (all-pass clip unverified)	Covered via #10
12	"Audiofrog Testgear UMI-1"	AudioFrog official channel	Nov 8, 2021	33:19	YouTube	Measurement-first REW workflow, mic calibration, measuring from outside, test-CD tracks	Yes
13	"How to Tune a Cadillac Deville ... PART2 Andy Wehmeyer to the Tune rescue"	361 VR	Jul 12, 2019	14:40	YouTube	Why tune, pink noise, mic arrays	Yes
14	"Andy Wehmeyer shows his new system at CES 2016"	Ricardo Rangel	Jan 6, 2016	5:31	YouTube	Multiseat demo car, real centre speaker	Yes
15	"How to Handle the 'Cheapest Option' Objection Like a Pro"	Mobile Electronics Association	Oct 14, 2024	1:04	YouTube	Price-shopper calls	Yes
16	"Talking car audio with audiofrog"	carsandcalibres	Mar 7, 2024	1:16	YouTube	Host narration only	Yes (no Andy content)
17	Erin's Audio Corner GS8ND2 review (Andy's written statement on the page)	Erin's Audio Corner	Sep 1, 2020	n/a	Page	What the shallow GS8ND2 is for	Text read

#	Title	Channel / host	Date	Length	URL	Best for	Transcribed?
18	"AUDIOFROG"	Ricardo Rangel (streamed or premiered)	about 2020	15:01	YouTube	Unknown; Andy's presence unconfirmed	No (page froze twice)
19	"AUDIOFROG CUSTOM SPEAKER INSTALLATIONS MADE EASY"	AudioFrog official channel	about 2015	1:58	YouTube	What comes in the speaker box	No (page froze twice)
20	"Andy From Audio frog demonstrating how to install GB 25"	Anthonys car audio life	Oct 4, 2022	2:04	YouTube	GB25 install (would have to be watched)	No (no captions)
21	"AUDIOFROG DEMO CAR VISITS OUR STORE!"	Absolute Electronix	Aug 3, 2023	6:03	YouTube	Volvo demo-car facts (7 amps, 3-way doors, 6"+1" centre); presenter not identifiable as Andy	Read; not counted as Andy

Not found as recordings: SEMA 2012 "Car Audio Mythbusters" seminar (listing); KnowledgeFest 2015 "Tuning Cars" Parts 1–3 with Larry Frederick (CE Outlook); his two MasterTech Expo 2023 seminars (car mapping; measurements and "audio truthiness"), previewed in #9. A TEDx talk by an "Andy Wehmeyer" (_e0by1vQVx4) is probably a different person. The official channel @audiofrog6034 has only two uploads, so his video teaching lives mostly in guest appearances.

Audiofrog Tech Blog (16 posts by Andy, Dec 2020 – Oct 2021; index)

1. Is it okay to use the terminal screw to crimp a ferrule? (Dec 15, 2020)
2. Tech Tip: Digital Audio Basics (Dec 21, 2020)
3. Pink Noise, White Noise and Why Your Tweeters never get 150 watts (Dec 21, 2020)
4. To Cap or Not to Cap? Tweeters in systems with active crossovers (Dec 21, 2020)
5. Time Alignment Part 1 (Dec 21, 2020)
6. Damping Factor and Why it isn't Much of a Factor (Dec 22, 2020)
7. Time Alignment Part 2 (Dec 22, 2020)
8. Time Alignment Part Three: Delays and Crossovers for Tweeters and Mids (Dec 22, 2020)
9. Crossovers: How They Work and How to Choose Them (Jan 14, 2021)
10. Measuring Frequency Response: Single Mic or Multi-Mic Average? (Jan 2021)
11. A Note about Target Curves (Mar 10, 2021; revised Apr 2021)
12. Are scopes and sine waves useful when setting up DSP? (Mar 24, 2021)
13. Testgear UMi-1 includes a CD with tracks designed for listening (Apr 14, 2021)
14. Dispersion and reflection (Jun 14, 2021)
15. How to test for polarity (Aug 17, 2021; revised May 2023)
16. Crossover Slopes and Protection for Speakers (Oct 7, 2021; revised Feb 2022)

(He also refers to a "Time Alignment Part 4" on sub/midbass crossover offsets; it is not in the list above and was not read.)

Testgear and other Audiofrog documents

- testgear.audiofrog.com: his free tuning guide, "A Straightforward One Seat Tuning Process and Some Notes About Why it Works" (about 65 pages; target-curve import on pp. 37–38), plus tuning-CD files and measurement notes. **Conflict:** in 2018 he described his stereo guide as two pages of process followed by about 60 of explanation; the round-1 notes describe about 40 pages of explanation followed by about 20 of process. His REW tuning document was written for REW 5.19 (DIYMA 2021-07-15); he also posted a Basic Delay Calculator spreadsheet (DIYMA 2018-05-22).
- UMI-1v2 tuning kit and the Binaural page; company pages for spec-reading: amplifiers, warranty, GB10OE tweeter sheet. Every Audiofrog sub manual includes sealed and vented box designs (B1 @7:18).
- His older "Tuning Cars" PowerPoint/PDF for retail installers (referenced in DIYMA 2016-01-29; not retrieved).

Magazine, press and retailer items

- Mobile Electronics: "Tech Today – DSP Tuning Part 3" (Aug 2017); all his columns via the Andy Wehmeyer tag page; Audiofrog items via the Audiofrog tag page. Best for tuning-as-troubleshooting and billing for tuning.
- EDUCAR: "What If There's More Than One Seat? Center Channels and Upmixing"; author page.
- CE Outlook via Big 5: "The Aftermarket Responds To Autoblog" (Jul 2018). BusinessWorld: "Demystifying good sound" (Mar 2024).
- Crutchfield: "How to choose the right Audiofrog components" and "Who is Audiofrog?".
- Otomotifnet/GridOto (Indonesian): profile, Oct 2016; Audiofrog U training, Aug 2019.
- Erin's Audio Corner reviews relaying his design notes: GB15, GB25.

DIYMobileAudio accounts and their most valuable threads

Both accounts show only anonymized numeric usernames.

- **"14642" (2008–2016, Harman and early Audiofrog years):**
  1. "JBL's MS-8 processor!" and the "JBL MS-8 FAQ" compilation: MS-8 logic, industry views, contest stories.
  2. "HOW COME NOBODY MAKES A GOOD DSP??" and "Please don't shut this thread" (2016): auto-tune limits, retail tuning process, judging speakers by data.
  3. "The more I look at Sinfoni" and "Are break-in periods necessary/ worthwhile?" (2015): reading graphs, QC, break-in.
  4. "Time Alignment Via Tape Measure Site..." and "Xover phase shift" (2015).
  5. "Definition of 'Sound Quality' inspired by IASCA" and "ultimate consequences of buying online" (2008): contests, OEM vs aftermarket, retail economics.
  6. Classic myth threads: "Damping Factor" (2011), "Lossy vs lossless formats" (2013), "Phase coherency: The Holy Grail of Car Audio" (2013), "HRTF" (2016).
  7. "Andy is Leaving Harman" (2013).
- **"156546" (Feb 2017 – Mar 2022, about 810 posts; his first post under this name says it was a new username):**

1. "Audiofrog Test Gear" (2017): one-seat tuning in two conditions, spatial vs single-mic averaging, MS-8 internals.
2. "Making stage wider" (2017): balance test, no images outside the triangle.
3. "Five Channel Soundstage" (2017) and "Is Rear Fill (and center channel) helpful or hurtful" (2018): how upmixers really work.
4. "Tuning Tricks" (2018): no tricks, only a process.
5. "RTA/REW- 90 degree or 0 file" and "Time delay measurements..." (2020): mics, averaging, LR4, reflection arithmetic, Harman-curve origin, patents.
6. "Clipping revisited with more evidence" (2020) and "Will too little power damage a speaker?" (2021): tweeter power arithmetic.
7. "Time Alignment revisited. Use technology!" and "Acoustical crossovers and volume" (2021): tape-measure delays, bass-up-front recipe.
8. "Polarity mystery, need some help" (2021): factory all-pass filters.
9. "House Curve – tweeters in A-pillar / woofer in lower doors" and "How to compensate for Comb Filters" (2021): measure-then-listen, target at the dips.
10. "Understanding DSP Output Voltage..." (2021): gain structure.
11. "Time to educate y'all bit depth sampling rate bitrate" (2021): digital audio, vinyl, Bluetooth.
12. "Speaker cone material", "Differential inputs from your headunit" and "Power handling, gimmick or..." (2021).
13. "Audiofrog manufacturer" (2020–21) and "Good Chinese speaker manufacturers on Alibaba" (2022): manufacturing and origin.
14. "Help: when to stop" (2022).

- **Not covered:** his daily tech tips on Facebook, Facebook SQ groups and the Audiofrog web forum.

Appendix B. How this research was done

- **Forum posts, account "14642" (2008–2016):** about 75 member-restricted keyword searches (about 2,200 result posts screened), the member-search pages 1–25 (about 500 posts, Apr 2015–Jun 2016) and the MS-8 FAQ compilation thread; only posts whose markup carried his author ID were used.
- **Forum posts, account "156546" (Feb 2017–Mar 2022):** every post in his member-search listing, about 810 in all (240 + 140 + 40 + 80 posts on the newest pages 1–25, and 240 + 71 on the older-date pages), each author-checked; one post (4829145, Jun 2017) could not be read, and two thread-opening posts lacked a captured author attribute but appear in his member list.
- **Audiofrog material:** all 16 bylined Tech Blog posts, the two tuning PDFs on testgear.audiofrog.com, and unbylined company pages (marked COMPANY, probably his but unsigned).
- **Video and audio:** 16 appearances transcribed from YouTube auto-captions (14 in round 3, plus the CMA Connected 2021 and Absolute Electronix 2023 livestreams in round 4); Erin's Audio Corner text; obvious caption mishearings (e.g., "audio fraud" for Audiofrog) were corrected, and timestamps are approximate.

- **Articles:** Mobile Electronics columns, EDUCAR, CE Outlook/Big 5, BusinessWorld, Crutchfield, Otomotifnet/GridOto, SEMA and CE Outlook listings.
- **Method:** his own words only (host and forum-member statements used only as labelled context); everything paraphrased, quotes under 15 words; RARE marks points that contradict or go well beyond common advice; duplicates across years merged, with the dates kept to show consistency or change. Apparent typos were reproduced and flagged in the notes (e.g., "22,500 Hz" for the 44.1 kHz Nyquist limit; Bluetooth "1kbps").
- **Known gaps, topics:** he barely addresses the "Big 3" upgrade (only that it rarely fixes noise and should be crimped), stiffening capacitors (only "not needed, remove it") and A2B digital audio buses (nothing found); whole-system wire-gauge budgets and deadening detail are thin in the 2017–2022 posts.
- **Known gaps, sources:** three clips untranscribed (Appendix A #18–20); his MasterTech 2023 and SEMA/KnowledgeFest seminars not found as recordings; the MEA channel, 12voltnews and podcast apps for 2024–2026 not searched; Facebook not covered; linked Dropbox charts, the old "Tuning Cars" deck and one Mobile Electronics Q&A page not retrieved.