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Crossovers and Biamplification

by

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Crossovers and Biamplification

An old technique that is recently regaining popularity in autosound and hi-fi circles is biamplification. There are quite a few good reasons for converting your system to biamplification. Among some of the more common ones are:

• Additional peak output capability

• Increased damping factor at woofer

• Lower intermodulation distortion

• Improved safety factor for drivers

• Ability to use more exotic crossover networks

• Ability to optimize the crossover frequency

• Biamplified systems play louder, cleaner and more reliably.

In this technical paper, we will explore the world of biamplification and as a consequence, discuss the ramifications of picking a particular crossover frequency. If you’re not completely at home with exactly what a crossover network does, refer to Technical Paper 102, “Crossover Networks from A to Linkwitz-Riley.”

What is Biamplification?

Biamplification is simply the technique of using a crossover network and individual power amplifiers for each frequency band required by the loudspeaker system. In its simplest form, a two-way system, two amplifier channels would be required: one for the woofer (low frequency or LF) and one for the tweeter (high frequency or HF). For stereo, four amplifier channels are required, since there are four loudspeakers involved (2 LF and 2 HF). Figure 1 shows the differences between a biamplified system and an conventional system.

Conventional (that is, loudspeakers with integral crossovers) systems also use a crossover, but the difference is that the crossover must operate at loudspeaker power levels and impedances. While this is permissible for a small bookshelf system used at home, the limitations imposed by more critical listening or higher operating levels make the internal or high level passive crossover impractical.

A loudspeaker crossover is nothing more than a collection of electrical filter circuits. Most filters operate best when terminated (loaded) by a pure resistance. A real-world loudspeaker is anything but a pure resistance. In the case of the woofer, there are one or two impedance peaks (one peak in a sealed box, two peaks in a vented or ported box). Tweeters and midranges are a bit better behaved, their largest problem being the increase in impedance with frequency caused by the inductance of the voice coil (woofers also have this problem). This change in impedance can be corrected, but the problem of the humped impedance curve of the woofer remains. By now, it should be clear that designing a passive crossover network is an exercise in compromise.

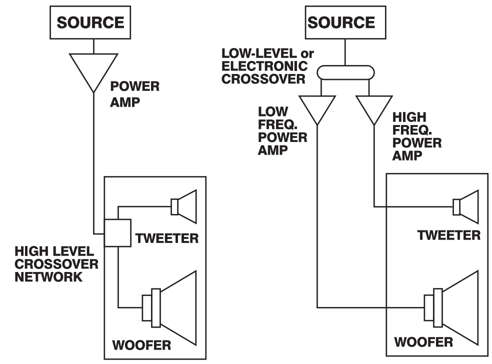

Figure 1. High level (passive) and low level (electronic) crossovers.

Biamplification isn't new. It's been with us for quite some time...easily since the days of vacuum tubes and mono. For many applications, the extra expense and complexity aren't worth the improvement. If you want the best, biamplification is de rigueur.

How does it work?

Consider a traditional two-way system: woofer and tweeter driven by a single 60 watt amplifier. The test signal is a jazz group consisting of piano, bass and drums. As we increase the volume level, the sound level in the room increases, up to a point, that point being where the power amplifier begins to clip. Why does it clip? Because the demands of the test signal have exceeded its power output capability.

If you look at the test signal on an oscilloscope, you'd see the piano and drums riding along on the bass waveform, occasionally punctuated by the impulsive waveform created by the bass (kick) drum. If it weren't for the bass information, you could probably increase the volume level even more. What happens is that the low frequencies demand all of the power that the amplifier has, leaving little or none for the high frequencies. Once the amplifier clips, everything going through it at that time is clipped. The loudspeaker simply does what it is told and tries it's best to reproduce the clipped waveform.

Next, consider a biamplified system: woofer driven by the same 60 watt amplifier used in the previous example and tweeter driven by a separate 20 watt power amplifier. The test signal is the same as in the previous example. As we increase the volume level, the sound level increases until the woofer amplifier clips. Increasing the volume level beyond this point results in the piano and high frequency components of the drums getting louder. Eventually the high frequency amplifier clips. Note that we were able to raise the volume level past the point where the woofer amplifier clipped.

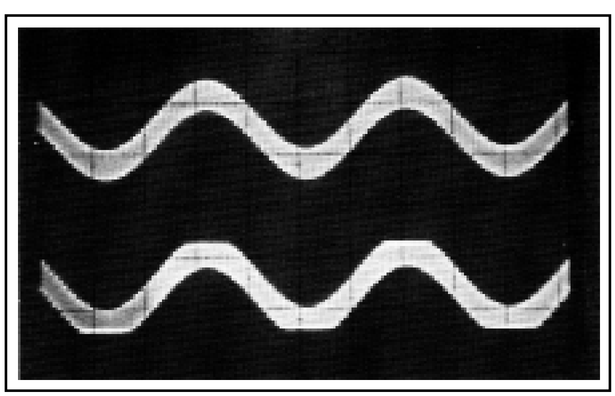

Figure 2. A strong low frequency signal causes high frequency clipping.

About now, you're probably wondering, "Why doesn't he just get an 80 watt amplifier and be done with it?" Because a biamplified system using a 60 watt and a 20 watt amplifier has a potential peak output capability of over 149 watts! Now before you run right out and do something rash, let me point out that this figure is very optimistic. For this line of thinking to hold true, the following things must occur.

- The signals driving the amplifiers must both cause their respective amplifiers to reach their peak output capability at the same instant in time. If the signals don't reach peak output simultaneously, then the 149 watt figure begins to shrink.
- The input signal driving the crossover/amplifier system must be spectrally diverse enough that both amplifiers are used to their peak output limitation. If the input signal contains little or no information above the crossover frequency, then the output capability is only 60 watts.
- The math that describes this exercise is described in the Appendix.

Why does it work?

Biamplification works because of the spectral distribution of music. Very simply stated, it works because the bass frequencies require the most power to reproduce. Now only is there more energy in the bass frequencies (bass instruments tend to be louder), but the loudspeakers used for bass reproduction tend to be less efficient than those used for treble reproduction. This double whammy really eats up amplifier power.

As the low frequency amplifier begins to clip, the output of the high frequency amplifier remains clean (unclipped). The low frequency clipping is masked by the unclipped high frequency output of the tweeter. The woofer further helps this since its frequency response is usually rolling off anyway (most woofers make lousy tweeters). Having high frequency output capability in excess of the woofer gives you extra high frequency headroom, which can subjectively make your system sound considerably louder. There is a limit, though. When the low frequency section clips, its output ceases to increase...it does not get any louder. Once the low frequency section enters clipping, the balance of the system begins to shift, getting thinner and thinner sounding.

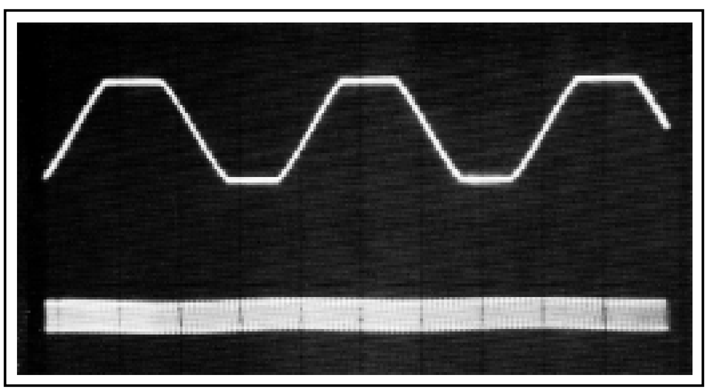

Figure 3. Clean high frequencies while low frequencies clip.

What situations can really benefit from biamplification?

- Systems requiring large amounts of peak headroom or dynamic range.
- Systems that need to extract the last 1dB of the component's capability.
- Systems using subwoofers
- Systems where there is a large difference in driver sensitivity.

How much amplifier power do I need?

The amount of amplifier power required depends on these factors:

- Driver sensitivity
- Sound pressure level requirements
- Frequency distribution of the source material

From a practical standpoint (maximum smoke), you probably need an amplifier with around 1.5 to 2 times the continuous power that your low frequency speakers can handle.

- The extra wattage helps keep the amplifier out of clipping, which imposes greater demands on the speaker than does the extra wattage. When am amplifier clips, it's output becomes a square wave (more or less). Once this happens, the energy distribution of the output only vaguely resembles the input signal. In this case, the energy delivered to the speaker approaches two times the sine wave output capability of the amplifier.
- The high frequency amplifier needs to be large enough to drive the high frequency speaker to perhaps 6dB louder than the low frequency speaker. If this figure is more than 1.5 to 2 times the power rating of the high frequency speaker, then your tweeter may have trouble keeping up with your woofer without self-destructing.

Time for an example:

We have a two-way system consisting of a single 15 inch woofer with a rated sensitivity of 93dB with an input signal of one watt, measured at one meter distance from the speaker (93dB/1W/1M), and a horn loaded tweeter with a sensitivity of 111dB/1W/1M. You can find these numbers on the spec sheets for many products. The woofer is rated at 300 watts and the tweeter is rated at 50 watts. Note that the tweeter is 18dB more efficient than the woofer. This works out to be 63 times more efficient than the woofer. Here's some other ways of looking at this:

- It takes 0.0159 watts input to the tweeter to equal the woofer's output at one watt.
- At full power input (50 watts), the tweeter is capable of over 127dB SPL while the woofer is only capable of producing 117dB at full power input (300 watts).

If we put a 300 watt amplifier on the woofer, how large an amplifier do we need for the tweeter in order for it to deliver 6dB more level than the woofer when it's amplifier is clipping? How much SPL will be delivered with the woofer amplifier at clipping?

1. Convert the amplifier power into dBW:

$$\begin{aligned} \text{dBW} &= 10\log(P) \\ &= 10\log(300) \\ &= 10*2.4771 \\ &= 24.711\text{dBW} \end{aligned}$$

2. Add this to the woofer's sensitivity:

$$93\text{dB} + 24.77\text{dB} = 117.77\text{dB}$$

This represents the woofer's output at 1 meter, in free space with the woofer amplifier at clipping.

3. Now calculate what 6dB past this level is:

$$117.77 + 6 = 123.77\text{dB}$$

4. Subtract this from the tweeter's sensitivity:

123.77 - 111 = 12.77dB

This represents how many dB over 1 watt you must drive the tweeter to be 6dB louder than the woofer.

5. Now figure out how many watts this is:

Power Ratio = *antilog* (12.77/10)
= a*ntilog* (1.277)
= 101.277
= 18.93 watts

since our ratio is against 1 watt, 18.93/1 = 18.93.

A power level of 18.93 watts into the tweeter results in an acoustical level of 123.77dB SPL at 1 meter. This represents a minimum figure; any amplifier power up to and including the tweeter's power rating will suffice.

In many cases you may want to exceed the transducer's (speaker's) power ratings. This is permissible, providing that you have a way of insuring that the high frequency amplifier NEVER clips (limiting, insufficient drive from the crossover, etc.). The average low frequency speaker can handle moderate amounts of clipping in stride, but the average high frequency speaker (tweeter) can't.

When an amplifier clips, the spectrum (frequency distribution) of its output changes considerably from that of the original signal. If the cause of the clipping is the inability of the amplifier to amplify two mixed signals, like the piano and drums in the previous example, the clipped combination of the two signals generates additional frequency components not present in either of the two original signals (intermodulation distortion). The direct consequence of clipping is a shift in the energy distribution of the signal resulting in low frequency components and additional energy in the clipped signal. Feeding this signal to a tweeter via a passive crossover is usually disastrous. Biamplified systems avoid this problem because the two signals are amplified separately.

Driver Protection

A passive high-level crossover provides protection against excursion damage for the midrange and high frequency drivers. A common failure mode of transistorized amplifiers is to have one or more output transistors fail, which has the effect of connecting the loudspeaker load directly across the power supply in the amplifier. This is usually the kiss of death for the poor speaker. In passively crossed-over systems, the crossover will pluck the tweeter and midrange speakers from the jaws of death.

In a biamplified system, it is good practice...no, make that insurance...to provide some sort of DC protection for the midrange and high frequency loudspeakers. In its simplest form, this is a series capacitor whose value is enough to form a simple crossover at 1/2 of the crossover frequency.

For example, I have a two-way system, crossed-over at 800Hz. I set the value of the protection capacitor for a 400Hz crossover. The formula used is:

$C = 1{,}000{,}000/(2\pi RF)$

where:

$\pi = 3.14$
R = the impedance of the driver, in ohms
F = the frequency in hertz
C = the value of the protection capacitor in microfarads

For the example, assuming an 8 ohm driver, the protection capacitor would have a value of 49.7 microfarads. The nearest standard value is 50 microfarads, which will work fine.

The best capacitors to use are non-polarized, non-electrolytic ones. The best ones currently available use a polypropylene dielectric. These are very expensive and even more difficult to find in high values. A more sensible approach is to use two polarized electrolytics in series. This means that you connect their plus leads together and use the two minus leads for your connections. You must buy capacitors with twice the calculated value to do this. You can still get most of the benefits of the polypropylene capacitors by paralleling the electrolytics with a single moderate value polypropylene unit, say 1 microfarad. Although this results in a larger overall valued capacitor (the values add), since this is just for protection the slight increase in value doesn't matter.

What about crossovers?

There are two general types of crossover networks in use today: active and passive. Used within their limitations, both are equivalent, given equivalent specs. As you will see, the passive network has very definite limitations.

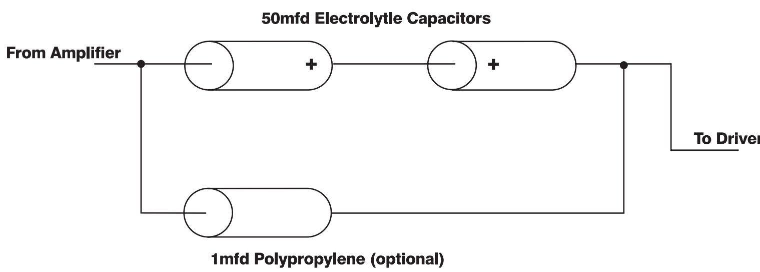

Figure 4. Protection capacitor connections

First, what is the difference between an active and passive crossover? Very simply stated, the passive crossover requires no external power source besides the audio signal itself. It does not depend on any sort of external power. An active crossover requires an external power source for operation (i.e. you must feed it batteries, or plug it in).

A second difference arises in terms of design. Passive crossovers require one reactive component per 6dB of slope. A reactive component is either an inductor (coil) or a capacitor. Active cross-overs use resistors and capacitors to achieve the same end result. Furthermore, when the crossover design begins to get complicated, say beyond 18dB/octave, active crossovers take the lead because they're much easier to design. You can literally get any slope you want...from 6dB/octave to 72dB/octave...and beyond.

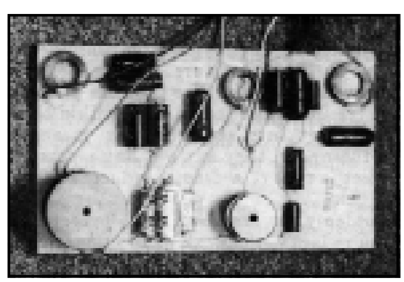

Figure 5. Active and passive crossovers.

Passive crossovers that are typically used in conjunction with loudspeakers operate best when presented with a resistive load, which a loudspeaker isn't. Most better designs attempt to compensate for the non-resistive nature of the loudspeaker load, but you can only do so much. Figure 6 shows the impedance curve of a typical 15 inch woofer in a ported enclosure.

In addition, the low frequency speaker has an inductor in series with it, which degrades the damping factor (ability of the amplifier to tell the speaker what to do) of the amplifier and causes small, but measurable amounts of signal loss. Figure 7 illustrates the effect of a varying load impedance on a typical passive crossover.

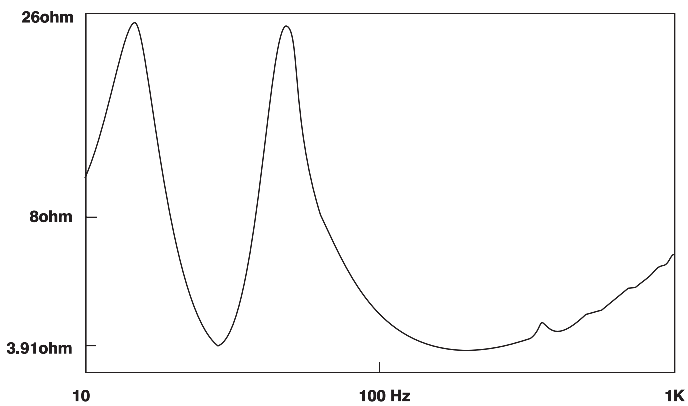

Figure 6. Impedance curve of a real world loudspeaker.

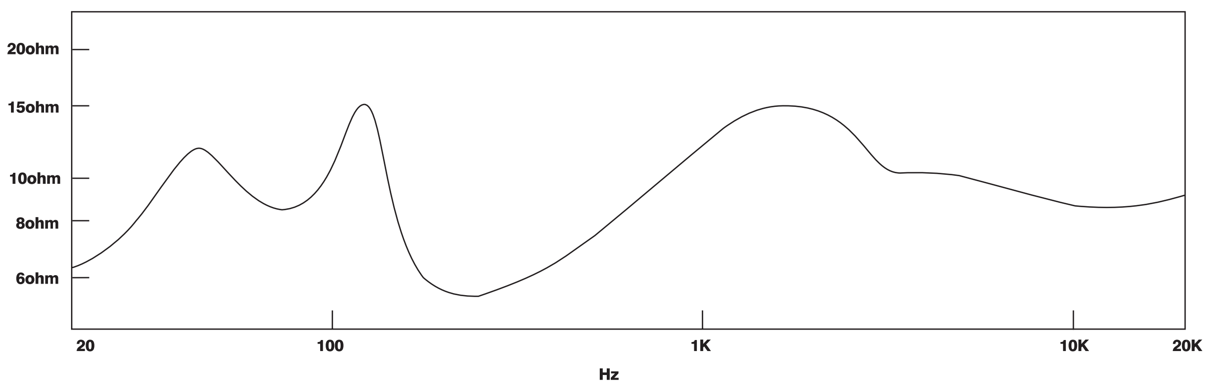

Figure 7. Variation in crossover characteristic caused by impedance variations.

Another problem arises when the amplifier used has much greater power capability than the crossover designer anticipated. This is very often the case when bookshelf speakers are used with large amplifiers in the quest if infinite headroom (playing the Telarc 1812 Overture at live performance volumes). Everything works fine at reasonable levels. The problem arises when a sudden peak exceeds the power rating of the inductors (coils) used within the crossover network. If the crossover uses iron core inductors (an economy measure, since it requires less wire for a given inductance), they have a finite power limit. The limit is core saturation and the effect is that of a drastic inductance change, which detunes the crossover.

Suddenly the crossover is no longer like the designer envisioned. All bets are off. If you're lucky, the peak passes unnoticed by the tweeter or midrange. If you're not...

A passive crossover isn't always used directly with a loudspeaker. Indeed, many of the earliest electronic crossovers were totally passive...textbook implementations of the appropriate network.

The difference here is the signal levels involved. A passive crossover connected between the amplifier and loudspeaker system is termed a **high-level crossover**. A passive crossover (or active crossover for that matter) used for biamplification is termed a **low-level crossover**.

An active crossover uses active filter technology to create the response characteristic of a similar passive filter network. Active crossovers are sometimes called electronic crossovers. Active crossovers have the following advantages:

- The crossover is isolated or buffered from the loudspeaker load by the power amplifier. Since the speakers are connected directly to the amplifiers, the damping factor is limited only by the speaker impedance and the wire resistance. The amplifier has better control of the speaker's cone motion.
- The insertion loss of the crossover network is no longer a factor.
- The power rating of the crossover is no longer a factor.
- It is much easier to tune to precisely the crossover frequency that you want.
- Steeper slopes are possible without the insertion loss of a high-slope passive crossover network.
- Intermodulation distortion caused by amplifier clipping is reduced.

Picking a crossover frequency

A consequence (or privilege) of using an active crossover is that you can easily tune the crossover to almost any crossover frequency. In some cases, you may have control over slope and other aspects of the filter characteristic. Changing a module to change the crossover frequency is trivial. Picking the right crossover frequency is anything but trivial.

The following factors influence your choice of crossover frequency:

- driver (loudspeaker) limitations
- polar pattern of the driver (dispersion)
- performance goals

Driver limitations

One of the purposes of a crossover is to allow the driver to operate in the optimum region of its frequency response. For midrange and high frequency drivers, the low frequency rolloff provides protection to the driver from excessive diaphragm excursion as well as preventing the driver from operating where it can produce little or no output.

In a similar vein, the high frequency rolloff of the bass and midrange crossover outputs prevents these drivers from operating in a region of their response curve where they may have irregular frequency response, or severe dispersion problems.

The consequences of operating a low frequency speaker at too high a crossover point are never fatal. The usual problem is simply that of a hole in the midrange because the low frequency speaker may not have much output in that region.

On the other hand, picking a midrange crossover point that is too low may be fatal to the midrange driver. There are two basic failure modes here: thermal and over-excursion. Thermal failure occurs because the low crossover point directs additional energy into the driver, which may not be equipped to handle it (the voice coil burns up). Over-excursion failure occurs because the lower crossover point requires additional cone movement from the driver. If you exceed the cone excursion limit of the driver, it will most likely fail.

High frequency crossover points have the same implications as the woofer-midrange crossover: too low and the tweeter may die from over-excursion or thermal failure, too high and there may be a hole or irregular frequency response caused by operating the midrange driver at too high a frequency.

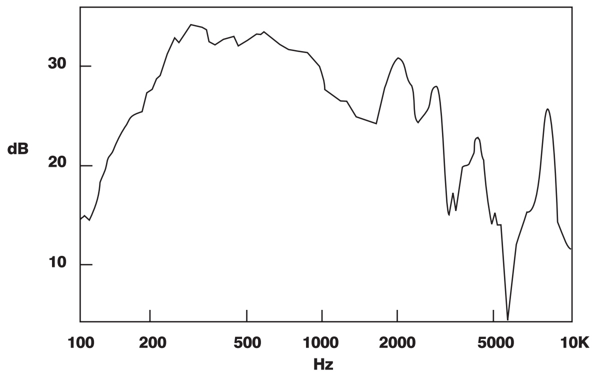

Figure 8. A typical midrange driver.

The Driver's Polar Pattern

The polar pattern of a driver is simply its dispersion characteristic, which is anything but consistent with frequency. Why does it matter?

It matters because the crossover allows the two loudspeakers operating throughout the crossover region (that is, the region where the two curves cross) to contribute simultaneously to the system's total acoustical output. Dispersion effects the ability of the system to project a stable stereo image over the sound stage. Irregular dispersion cause imaging problems such as

- the inability to localize certain instruments
- inconsistent localization throughout the listening area
- frequency response problems in the crossover region

The optimum crossover point is that where the dispersion characteristics of the two drivers involved match. If you are not an equipment manufacturer, this may be difficult to do. Remember, the limitations of the drivers come first.

Performance Goals

The performance goals of the completed system may not effect the actual crossover frequency so much as they may effect the number of crossover points. This is not to say that more is better. A well designed two-way system can easily hold its own against three-, four-, and five-way systems.

Generally, getting more performance out of a system means obtaining drivers whose performance characteristics are optimized over a wider range. It can also mean buying more drivers, and operating them over a smaller portion of their operating characteristic. This is the idea behind most multi-way (three-way and up) systems. The multi-way system is not a panacea for poor design...the design can be more difficult because of the multiple crossover points (phase interaction, lobing errors, time offset correction, etc.).

For some systems, performance goals do influence the choice of crossover frequency. In horn loaded systems, especially those where the high frequency portion of the system is horn loaded (such as a traditional "theatre" system) you generally want the crossover point to be as low as possible to take advantage of the horn loading on the driver. The limiting factors are:

- the low frequency limitations of the high frequency driver (exceed them and it will die on you)
- how loud you want the thing to go (the lower crossover point restricts the maximum output because the diaphragm must move further. . . it can only move so far)
- the low frequency limit of the horn (it can no longer load the driver below a certain frequency)

Subwoofers

Many folks discover biamplification and electronic crossovers by adding a subwoofer to their system either for additional low frequency output, or for extending the low frequency output of their system. Since the lower frequencies (below 250Hz) are fairly omnidirectional, it has become common practice to simply sum the lower frequencies and use a single speaker for the low bass region. The omnidirectionality makes placement fairly uncritical. Even if you don't have a subwoofer, you may still be enjoying mono bass because most disc recording engineers reduce the separation of the stereo signal to zero at low frequencies, which makes them mono. The reason for doing this is that it makes it easier to physically cut the record; large out-of-phase low frequency signals can cause skipping.

The crossover frequency for your subwoofer should be high enough to relieve the existing system of the burden of reproducing the low bass frequencies, but not high enough that the directionality of the subwoofer becomes a problem. In practice, 90 to 150Hz is a good range to try. Crossing over below 90Hz doesn't buy you much if anything; there really isn't that much information down there (unless you use a bass restoration unit) and the existing system must reproduce the 80-120Hz region, which contains a great deal of the bass energy on commercially recorded product.

Last, remember that the usual reason to add a subwoofer is to extend the lower frequency limit of your system. This does not mean you have more bass, but the bass that you have will go lower in frequency. The other benefit (if you biamplify) is the ability to alter the balance of the subwoofer with the rest of your system. Deliberately increasing the subwoofer level can give you the subjective quality of more bass.

Wrap-up

Remember the following rules when you biamplify your system:

1. Pick your drivers based on your performance goals.
2. Identify the individual driver limitations and actual performance characteristics.
3. Pick your crossover point, based on driver limitations, their polar patterns, and your performance goals, in that order.
4. Pick your power amplifiers, based on driver limitations, and your output requirements.
5. Incorporate DC protection for the midrange and tweeter elements of your system. The tweeter you save may be your own.

Biamplification (or triamplification) is a logical step for any high performance audio system. Well designed systems can play louder, for longer periods of time, without listening fatigue. If you are searching for an answer to the rigors of digitally recorded source material, biamplification may be the answer, providing that your speakers are up to the task. Now, sit back and listen to the music.

Appendix

By now, you're probably wondering how we made 80 watts equal 149 watts. Here's how it works.

The effective output capability of a biamplified system is the sum of the peak output voltage of the two amplifiers. We mentioned earlier that this was an optimistic figure. It is.

The reason that it is optimistic is that the effective output capability of a system (compared to one that uses a passive, high-level crossover) depends on a low frequency signal that is just barely strong enough to cause the amplifier to clip and a high frequency signal that is just barely strong enough to cause the high frequency amplifier to clip. The two signals must lie on opposite sides of the crossover frequency. Furthermore, the high frequency waveform needs to reach its maximum about the same time that the low frequency signal does (not as hard as it sounds because of wavelength differences).

Another factor that helps make this work is when there is a large difference in the sensitivities of the loudspeakers used. Let's use the speakers used earlier in our example. If we used these in a system that uses a passive crossover, then we would need to add 18dB of attenuation to make the sensitivity of the high frequency speaker match that of the low frequency speaker. What happens to the 18dB of power that we lose in the attenuator (the attenuator is typically the L-pad on the rear of the speaker)? It turns into heat. Remember that watts cost dollars!

In a biamplified system, all of the power generated by the amplifiers is seen by the speakers. If you have to reduce the output of one part of the system to match the other, you do it ahead of the power amplifier by using the level controls on the crossover or by using the power amp's input level controls. This does not waste power. Assuming that your preamp has enough output capability to overcome the loss, the power amplifier connected to the crossover can still reach full output. You can't say that about an amplifier that is driving an 18dB attenuator.

What does this mean? It means that (using our example speakers again), that a 50 watt amplifier will drive the high frequency speaker to 127dB SPL (loud!). The same speakers with a passive crossover would require a 3150 watt amplifier to be able to deliver 50 watts to the high frequency speaker! Doesn't it make a lot more sense to biamplify the system and only use a 50 watt amplifier for the high frequency speaker?

Okay, here's how you accomplish this bit of numeric magic. We'll use the amplifiers that were used in the example.

First, convert the amplifier's output wattage into an RMS voltage:

$$E_{rms} = \sqrt{P * Z}$$

where P is the amplifier power at load impedance Z.

$$E_{rms} = \sqrt{60 * 8}$$
$$E_{rms} = 21.9\,volts$$

Now convert RMS volts into peak volts by multiplying by 1.414.

$$E_{peak} = 1.141 * E_{rms}$$
$$E_{peak} = 1.141 * 21.9$$
$$E_{peak} = 30.98\,volts$$

Make the same calculation for the high frequency amplifier

$$E_{peak}=1.414*\sqrt{P*Z}$$
$$E_{peak}=1.414*\sqrt{20*8}$$
$$E_{peak}=17.89 volts$$

Next, add the two voltages. This represents what happens when a high frequency signal rides on top of a low frequency signal and both reach maximum simultaneously.

$$E_{totalpk}=E_{lf}+E_{hf}$$
$$E_{totalpk}=30.98+17.89$$
$$E_{totalpk}=48.87$$

Now convert the peak voltage back to RMS by multiplying by 0.707.

$$E_{totalrms}=E_{totalpk}*0.707$$
$$E_{totalrms}=48.87*0.707$$
$$E_{totalrms}=34.55 volts$$

Finally, convert this voltage into watts:

$$P=E^2/Z$$
$$Pequiv=E^2_{totalrms}/Z$$
$$Pequiv=34.55^2/8$$
$$Pequiv=149.22 watts$$

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