1.0 - The Basics of Bi-Amplification

Bi-amping is a technique which uses one amplifier for the low frequencies, and another for mid and high frequencies. The choice of crossover frequency is not too critical, provided that the amplifier powers are properly balanced to achieve the maximum benefit, and the drivers used are operating well within their frequency and power limits.


Figure 1 - Bi-Amplification Block Diagram

Figure 1 shows the basic concept in block diagram form. Only one channel of the stereo pair is shown, the remaining channel is identical. Note that the midrange to high frequency crossover retains a passive design - more on this later in the article.

In a simple form (using really simple electronic crossovers and little amps) biamping can be used even for computer speakers, clock radios and the like. The cost of the little amps is low, and the sonic improvement can be quite dramatic. I used to have a sub-woofer on my clock radio (really) and it actually sounded quite decent - at least insofar as a clock radio*can* sound decent.

As a solution to just about any amplifier-speaker combination, biamping has to be the way to go. At the highest or lowest ends of the audio equipment range, a biamped system will sound better than conventional passive crossovers, and one amp doing all the work.

The Most Common Question About Biamping

The most common question I get is ...

"Do I need to disconnect the passive crossover in my speakers?"
The answer is ... Yes, otherwise you are not really biamping at all.

Generally speaking, the mid to high section needs to be retained since a typical biamp setup will only eliminate the bass to mid+high network. These sections are nearly always completely separate networks, although it may not seem like it when you first have a look at the board.

Equally important is the selection of the electronic crossover frequency. It must be the same as the original, within a few 10s of hertz. The only exception is where you might obtain information from the manufacturer of the speaker that allows the frequency to be modified. In general, I strongly suggest that you determine the original crossover frequency, and stay with it.

When the crossover is modified, make sure that you retain all the parts, along with the original connections. A drawing (including all component values) and photograph will be of great assistance when you want to restore the speakers to normal prior to selling them - it is unlikely that you will ever want to do this for your own use - not after you have enjoyed the benefits of biamping for any length of time.

Passive biamping (where two amplifiers are used in a bi-wiring connection) is, IMHO, a waste of money. Although there may be some moderate sonic benefits, they are not worth the expense of the extra amplifier.

1.1 - Terminology

In writing this article I have endeavoured to keep technical terms to a minimum. Unfortunately, this is quite impossible (for me anyway), so if you are not familiar with the terminology used, please refer to the [Glossary of Terms](http://sound.westhost.com/glossary.htm), now in a separate page.

This includes some terms which are not in the body of the article, but are useful nonetheless since they may be encountered elsewhere.

1.2 - Speaker Sensitivity

Initially, let us look at an "ideal" situation, where the loudspeakers for low and midrange plus high frequencies have the same sensitivity (say 90dB / watt @ 1 metre). This means that in free space (without reflections from walls etc.), the speaker will provide an output of 90dB SPL (Sound Pressure Level) measured at a distance of 1 metre with an input power of 1 Watt. This will usually be measured with band limited noise, so the speaker's little peaks and dips will not overly influence the measurement.

The high frequency driver (tweeter) is of minimal interest at this point in the discussion, so will simply be lumped in with the midrange to give mid+high.

1.3 - Power Distribution and SPL

With typical program material (whatever that is), it has been determined that the "equal power" frequency between low and mid+high is between 250Hz and 350Hz. This is defined as the frequency where the bass and mid+high amplifier power requirements are equal. So with our 90dB/Watt/Metre speakers above we could assume that 100W amplifiers might be appropriate.

This will allow an absolute maximum of just over 110dB at one metre. You may think that is loud (you would probably be right, too), but this is the peak single-frequency SPL, and allows for transient signals - ensuring that at no time does the amplifier clip (cut-off the tops or bottoms of the waveform). This introduces distortion which quite apart from sounding awful, causes listener fatigue and places loudspeaker drivers at risk of damage.

The [Speaker Damage](http://sound.westhost.com/bi-amp-p1.htm) popup has more information on this topic for those who are interested. In addition, it is suggested that you look at the article [Why Do Tweeters Blow When Amplifiers Distort?](http://sound.westhost.com/tweeters.htm) for further details

The actual (averaged) SPL at one metre will be somewhere in the vicinity of 90 to 100dB, depending upon the program material. The average SPL at the listening position cannot be determined without complete analysis of the room's acoustics (for a typical room you will lose another 6 to 10dB), so for simplicity we will use the 1 metre SPL as a reference value.

Thanks to a reader, here is a small table that shows the power distribution at different crossover frequencies. The table came from a loudspeaker manual "LOUDSPEAKER ENCLOSURE DESIGN AND CONSTRUCTION" published by FANE.

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| --- | --- | --- |
| X-over Frequency (Hz) | Power to Bass (%) | Power to Mid+High (%) |
| 250 | 40 | 60 |
| 350 | 50 | 50 |
| 500 | 60 | 40 |
| 1,200 | 65 | 35 |
| 3,000 | 85 | 15 |
| 5,000 | 90 | 10 |

Note that according to this table, the equal power point is 350Hz (which I calculated, since it was left out of the original). This is slightly different from my own measurements, but the error is of no consequence, regardless of who is right. As can be seen, the power requirement falls quite rapidly after 1200Hz, and although not shown, it also falls off with reducing frequency.

Since the last statement will possibly cause some discomfort or indeed confusion (after all, everyone knows that a subwoofer needs more power than the main speakers), I should explain myself. Most of the time in this article, I refer to power as average power, and indeed the average power falls with frequency below about 100Hz or so. The peak power is a different matter, and depends to a very great degree on the type of music.

The table assumes equal efficiencies for the bass and mid+high drivers. Should they be different, then a correction factor must be added in. For example, if the bass driver were to be 3dB less efficient than the mid+high drivers, then the bass power must be doubled (and of course vice versa). If the difference is less than 3dB, you may safely double the power anyway, or calculate the actual power needed - this I shall leave as an exercise for the reader.

This is a very important point, and cannot be over-emphasised. Some subs (particularly those using the "Extended Low Frequency" ELF™ technique) will need a huge amount of additional power at the bottom end because of the way they are driven. It is not easy to give a simple formula (so I'm not going to :-) to calculate the power needed, because there are so many variables.

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| NOTE | Do not be tempted to reduce bass power below about the 40% level regardless of crossover frequency, because although the average power might be quite low, it is usually of relatively high peak amplitude. The wide dynamics of the bass content require an amplifier capable of far more power than might be imagined if clipping is to be avoided. Clipping is something that one should avoid at all costs, because apart from sounding horrible, the average power level is increased, placing loudspeakers at risk. Having said that, *some* peak clipping in a subwoofer may be inaudible, provided the remainder of the signal is clean.In general, I suggest that the bass amplifier should have at least the same power as that used for the mid+high frequencies, but if any equalisation is used (such as the Project 71 Linkwitz Transform circuit), this may need to be increased dramatically. A boost of only 6dB may require that amp power be increased by 4 times. |

1.4 - Actual vs Effective Power

If we assume that our 100 Watt amplifiers will be handling exactly the same peak amplitudes with typical program input, then we have a total of 200 Watts for the combined program material. So, where does the magic come into this? This amp combination will sound (and measure) as if it were 400 Watts - twice as much "effective" power as there is real power. For this to make sense, we need to back track a little.

Imagine a sine wave signal of 100Hz at an amplitude of 28V RMS. For an 8 ohm load, this equates to about 100W (98 actually). The same amplitude at 1000Hz will be exactly the same power. Now add the two signals together, in the same way that signals add together in music. We are interested only in the peak amplitude, the RMS value indicates that the power is only 3dB higher, but it is only when an oscilloscope is used that the true picture emerges.

We will now see a low-frequency waveform, with a higher frequency waveform superimposed - the high frequency signal will be riding up and down the path of the low frequency signal. If we were to perform a calculation (or simply measure the combined signal with an oscilloscope), we will see that the peak amplitude has doubled. The effective RMS value (most multimeters will get this wrong unless they are true RMS types) is 40 Volts, and this would imply 200W. Although this is the real RMS voltage, it totally underestimates the amplifier power needed to reproduce it cleanly. An oscilloscope shows 80V peak for the same waveform, so the amplifier must be capable of passing an 80V peak signal - a 400W amplifier.


Figure 2 - Addition of Waveforms

To illustrate this point, Figure 2 shows two signals, each of 1 Unit peak amplitude. As can be seen, when the two are combined, the amplitude is much greater. The maximum peak amplitude is now 2 Units - double the peak voltage and four times the peak power of each signal individually. Power increases as the square of voltage, so twice the (peak) voltage is four times the power. Real ('RMS') power increases by 3dB or double the power, but this is a misleading figure and cannot be used. An oscilloscope is essential.

*Note: Peak-to-peak amplitude is actually double the values quoted above, but since amplifiers are generally symmetrical (capable of equal positive and negative voltage swings) it is more convenient to simply refer to the peak amplitude only.*

This is not to say that the actual music will be symmetrical. It isn't, but it is completely unpredictable in nature. As a result, it is possible (for example) to set up an amplifier asymmetrically and adjust the phase to suit with a switching circuit, since it will change. AM radio actually does this (or they used to) - a circuit is used to switch the phase so that slight over modulation causes more transmitter power, but never reduces it below the acceptable minimum. I shall not be going into details, since I believe few audiophiles would find this acceptable - I know I wouldn't.

All signal sources have the same characteristics as shown above in Figure 2, even a solo voice or musical instrument. In these cases, the fundamental frequency forms the low frequency component, while the harmonics 'ride the wave' as it were. Not surprisingly, the 'equal power' frequency will change (often dramatically) from the 250 to 350Hz range quoted above, but the basic principle does not alter.

Completely beside the point (but interesting anyway) is that in many musical instruments, the harmonics are actually at a greater amplitude than the fundamental. (File this away under 'Useless Information'.)

Note: It must be explained here that the 3dB effective power increase is the absolute maximum that can be obtained. In most cases it will be less - I have examined sections of music where the power gain was less than 1dB, and it can be reasonably safely assumed that the real gain will lie somewhere between 1-2dB in most cases. The real figure depends a lot on the type of music, the actual crossover frequency, and the peak to average ratio of the two separated signals. Just this topic alone is sufficient for a complete article in its own right.

1.5 - Separating The Signals

A passive crossover will separate the two signals shown above and feed each to the appropriate loudspeaker in the system. The amplifier must be capable of handling the entire composite waveform, so for our previous example of 100 Watts for each signal individually, must be capable of 400 Watts to reproduce the waveform without distortion.

If we now we separate these signals again - prior to the power amp - and using an electronic crossover, we have an entirely different situation. (Note: It is assumed for the sake of this article that the crossover frequency is near the halfway point between the two discrete frequencies of Figure 2.) Each signal is now supplied to its own 100W amplifier (there will be but a hint of the other frequency still visible on an oscilloscope, since the filters are not "perfect") and thence to the loudspeakers. The amplifiers are not clipping, both signals are reproduced at their original power, and the effective result is that we are emulating a 400 Watt amp with two 100 Watt units.

By way of comparison, the waveforms in Figures 3A, 3B and 3C show what happens if the composite waveform is fed into a single 100 Watt amplifier, and we try to obtain the same power output as before. Once the amplifier's output voltage attempts to exceed the internal power supply voltage, the amplifier clips the tops and bottoms of the waveform - resulting in harsh distortion and placing tweeters at extreme risk due to the additional high frequency energy which is created by the sharp transitions of the clipped waveform, and even more so by the compression of the signal (see [Speaker Damage](http://sound.westhost.com/bi-amp-p1.htm)). This also adds a considerable amount of intermodulation distortion to the signal, so the distortion is not just harmonic, but can also be discordant (not harmonically related). This is the worst kind of distortion, and sounds really gross.


Figure 3A - Unclipped Waveform Expected from 400W Amplifier

Using the same principle outlined above, we add a 200Hz signal and a 2kHz signal, each having 1 unit (1V) amplitude. The result is a combined signal with a peak amplitude of 2 units. Again, if we equate 1 unit (1V) with a nominal 100W, then 2V is 400W.


Figure 3B - Clipped Waveform From Underpowered Amplifier

The result of feeding a 2V input signal into an amplifier that is capable of reproducing 1V is shown in Figure 3B. The waveform in Fig 3B is *exactly* the same as that in Figure 3A, except the amplifier has limited the peak amplitude to ±1V, so causing the signal to be clipped. It is not immediately apparent, but both the low and high frequencies are distorted by the clipping action, and it is obvious that a significant part of the signal detail is no longer available as it has been 'clipped' off.


Figure 3C - Spectrum of Clipped Waveform

It is quite obvious that some of the signal has gone missing because of clipping. Not so obvious is that additional new frequencies are created, and this is shown in Figure 3C. This is a spectrum of the clipped waveform. The normal (unclipped) spectrum simply shows two peaks - one at 200Hz and another at 2kHz, with both being exactly 1V in amplitude.

Figure 3C shows that there is a multiplicity of 'new' frequencies. The original frequencies are each reduced to 714mV, and new frequencies are added. We have simple distortion, adding 600Hz to the signal, as well as sum and difference frequencies. These add 1.8kHz and 2.2kHz (at over 10% distortion level), as well as 3.8kHz and 4.2kHz. 6kHz is added (the third harmonic of 2kHz at the same level as the new 600Hz signal), plus 7.8kHz and 8.2kHz. All of these signals are above 10mV in amplitude (1% distortion referred to 1V), but there are a great many more new frequencies below that level. The end result is a very harsh noise - it no longer qualifies as wanted sound (or music).

While it may seem that a bi-amped system gives you 'something for nothing' in the power department, this is not really the case. Four 100 Watt amps (2 x stereo 100W / channel) are going to be about the same price as (or perhaps more than) two 400 Watt amps (1 x stereo 400W / channel), but they will not be as highly stressed by high voltages, will probably run cooler, and each only has to handle a more limited frequency range. (For more good ideas on this concept, see [Summary](http://sound.westhost.com/bi-amp.htm#summary), below.). An electronic crossover is also needed, and this adds to the total cost of the system. Of course, the low frequency passive crossover isn't needed, so this offsets the overall cost somewhat. While you never get something for nothing, biamping probably comes as close as you'll get.

1.6 - High Frequency Energy Content

The basic principles described above also apply to the way high frequency signals are superimposed upon the low and middle signals. The main difference is in the energy (power) of the respective frequency bands. There is normally a relatively high amount of energy in the midrange band (see [Crossover Frequency Selection](http://sound.westhost.com/bi-amp2.htm#freq_sel) in Part 2) as well as in the low frequency band. However, as the frequency increases beyond the upper fundamental frequencies of most musical instruments, the amount of energy falls off. Typically this will occur from about 800Hz and up (but will vary widely depending upon the type of program material), and the energy content will be seen to drop at a rate of about 3dB per octave (and more rapidly again above about 5kHz).

Since with a 3-way system the midrange to tweeter crossover frequency will be at perhaps 2500Hz or so, we can expect that the energy content of the high frequency band will be 9dB to 12dB down compared with the low and mid ranges. If we translate that back to our original 100 Watt amplifiers, this equates to somewhere between 7 and 12 Watts (peak) - giving an average power of around 1 Watt.

Because the high frequency energy content is such a low value (about 1/10th that of the midrange band), there is not a lot to be gained by using another electronic crossover network to separate this from the midrange signals. If the goal is to obtain the absolute maximum SPL (such as for sound reinforcement) it will be well worth the effort, but for hi-fi the law of diminishing returns indicates that it is not generally worthwhile. However, for optimum clarity, there is no comparison. An electronic crossover is not affected by driver impedance, and is (comparatively) infinitely stable.

One area where it is certainly better to use the additional electronic crossover and more amps for the tweeters, is where the sensitivities of the midrange driver and tweeter are more than 2 or 3dB different. In this case, using a separate amp will allow the levels to be matched far more easily, and will eliminate the use of resistive pads in the crossover network. There is also the potential for a useful reduction in intermodulation distortion, although with good quality modern amplifiers this is normally very low.

1.7 - Intermodulation Distortion

Intermodulation distortion in an amplifier is a form of distortion created when two different frequencies are being amplified simultaneously. The effects of intermod are most noticeable when one of the frequencies is much lower than the other, and the high frequency signal is actually modulated by the low frequency. This is quite different from the signals simply adding as they are supposed to. The effect (musically speaking) is that the sound is muddied, and the highs lose their transparency. Individual instruments become difficult to separate as their harmonics all start to blend into a 'wall of sound' (have another look at Figure 3B - this is intermodulation distortion at its worst).

As described above, intermodulation distortion is not harmonically related, so its effect is worse than simple harmonic distortion. Transient Intermodulation Distortion (TIM or TID) is (supposedly) created when fast transients exceed the amplifier's ability to change its output voltage fast enough. Although uncommon in modern amplifiers, TIM is still theoretically possible, although it is very rare to find any programme material that will cause *any* reasonably competent amplifier any stress.

These effects can be hard to quantify, but by using two (or more) amplifiers, any problems will be greatly reduced.

By separating the low and mid+high frequencies from each other prior to the power amplifiers, we reduce (to a large degree) one of the major sources of intermodulation. This is a great benefit to the music lover, since the sound instantly becomes more open and cleaner.

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| +  | So far we have identified two major plusses - effectively more power, meaning that transients are less likely to cause amplifier overload (clipping), and reduced intermodulation distortion. But wait, there's more .... (No, sorry, we're completely out of steak knives!) |

1.8 - Passive Crossovers

For those who are unfamiliar with the setup of a three-way passive crossover, please refer to [Figure 5](http://sound.westhost.com/bi-amp.htm#fig5), which shows (and the text explains) the connections. The diagram shown is for a "bi-wired" system, but includes the conventional connections.

When an amplifier reproduces the entire musical range, coils (inductors) and capacitors are used in the speaker cabinet to separate the high and low frequencies so that each may be supplied to the appropriate loudspeaker driver. A loudspeaker can be a difficult load for any amplifier, but when additional inductance and capacitance enter the equation, this only makes matters worse. Add to this the fact that all passive crossovers introduce some degree of loss (in some cases as much as 3dB - which means that they are "stealing" half the available power), and one can see that getting rid of them cannot be such a bad thing.

Look at the impedance graphs for almost any speaker system, and it will be seen that there is almost always a dip in impedance (sometimes severe) at the crossover frequency. This is caused by the interactions of the loudspeakers and their inductor/capacitor networks, and in some cases can cause amplifiers considerable grief - especially at high power levels. Although few amps will fail, one can expect a reduction in effective output power as the protection circuits limit the maximum power available due to the loading of the crossover network.

These vague thoughts are brought into stark reality when one learns that the inductors and capacitors needed for the low frequency crossover are quite large values, which leads many speaker designers to compromise in the interests of economy. The inductors may have an iron or ferrite core - which improves its inductance, but ruins its linearity. So now the crossover behaves differently depending upon the amplitude of the signal. High value high quality capacitors are expensive, so again, bi-polar electrolytics are often used. It is often stated that these sound awful, although this is a somewhat contentious issue, but without any doubt their characteristics change with temperature and age. They also have rather mediocre accuracy against their claimed value (+20/-50% is typical), so a 10uF crossover cap may be 12uF, or as low as 5uF. This does not make for an accurate crossover network, and most reputable speaker manufacturers will not make this sort of compromise, at least not for their top-of-the-line models.

In addition, at high powers, the impedance of the voice coil rises because of the temperature rise in the voice coil. This is not stable, and varies widely with the music. So with loud passages, the voice coil temperature might rise significantly, which will severely impact the performance of the crossover - relying as it does on the load impedance being a constant. A loud bass solo followed by a relatively quiet but complex passage might create an interesting shift in the crossover frequency and phase response as the voice coil cools, which is unlikely to enhance the listening experience.

1.9 - Electronic Crossovers

In contrast, the electronic crossover uses active filters in the low-level signal path. These suffer none of the power "eating" problems of the passive variety, and are far more easily tuned to exactly match each other - both within the same unit (between low and mid+high), and from one unit to the next. Indeed, it is so easy to tune an electronic crossover that they could be (and often are) set up individually for the exact loudspeaker drivers installed in a given cabinet.

In the sound reinforcement industry, crossover frequencies may (or should) be changed to suit the type of music, or even to suit the acoustics of a particular venue.

There are no issues with the crossover frequency shifting, since it is stable and not at all reliant on the voice coil impedance. This will still change with the power level, but the effects are unlikely (certainly less likely) to be audible.

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| +  | Now there are another two plusses to add to the list - elimination of the low frequency passive crossover, resplendent with its inherent losses, potentially poor linearity and crossover point inaccuracy (either as manufactured or with time, or both), and the reduction of the difficulty of the load presented to the power amplifier. Both of these result in more effective available power, ensuring that transients are preserved and overall linearity is improved markedly. |

1.10 - Speaker Sensitivity

This rated a scanty mention above, but is a highly contentious issue and can cause (does cause?) many a fine speaker system to suffer from a relatively low overall sensitivity. When a speaker manufacturer chooses drivers for an enclosure, they should be the very best available for the intended final product. In many cases, although other characteristics may be ideal, the chosen drivers will have different sensitivities. This is generally solved by "padding", using resistive dividers to reduce the sensitivity of the more sensitive driver to match that of the least efficient. So if our hypothetical drivers (as described above) were to have the following efficiencies:

Low 90dB / W @ 1m

Mid 93dB / W @ 1m

High 92dB / W @ 1m

It is immediately apparent that the midrange loudspeaker requires only half the power of the low frequency unit for the same output SPL. (It is 3dB more efficient, and 3dB equates to half (or double) the power.) This will never do for a quality unit, so it must be padded back by 3dB if the midrange is not to be prominent. Likewise, the efficiency of the tweeter is also too high, so this must be padded by 2dB to bring it into line with the others.

This represents a whole bunch of power being thrown away, simply dissipated as heat in resistors in the crossover network. But wait! Amplifier damping factor is a much quoted and highly sought after commodity. It mainly affects the low frequency drivers, but midrange loudspeakers are just as likely to have their own little resonances, too. Admittedly, these are much easier to control than the low frequency nasties, but it does seem to be such a shame to use all that expensive cable to ensure the best possible response and damping factor, and there it is - gone - filched by a couple of grotty little resistors.

Oh, and speaking of resistance. Remember the inductor for the low frequency crossover? Well the resistance of that is probably between 3 and 20 times greater than the resistance of your expensive cable. Don't assume that bi-wiring helps this either, because it doesn't (more on that a little later).

With our bi-amplified model, we can simply adjust the relative gain of the amplifiers (and their power too, if maximum SPL is the goal) to bring everything back into balance. No power is lost as heat in redundant passive components, and we can ensure that the damping factor of both low and midrange drivers is not compromised by the crossover components. The low frequency loudspeaker in particular is driven directly by the power amplifier with only the speaker cable in between.

There are also some loudspeaker drivers that, for various reasons, will sound better if driven from a finite impedance. This may be to correct the Thiele-Small parameters, or (as has been suggested by one reader who referred me to a web site - in Russian!) to reduce driver intermodulation distortion. This is an area that I shall investigate when time permits, but I have been using this technique for many years, but for different reasons.

For more information on matching the amplifier power to the speakers, see [Correcting Crossover Filter Amplitude Response](http://sound.westhost.com/bi-amp.htm#xover-gain), below.

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| +  | Add two more plusses. No padding is required to align the driver sensitivities, so we are not simply wasting power, and the damping factor is greatly improved for both the low and midrange loudspeakers (or can be individually set to the impedance that makes the speakers the happiest). |

1.11 - Phase Response

This one is nearly as big a "killer" as the power gain - and from a musical point of view it may well be seen as even more important (I happen to think it is). The phase response of any crossover is quite predictable, as long as the source and load impedances are well defined and stable. In a passive crossover, this is rarely the case, and the results can be quite nasty. There is a phase transition around the crossover frequency, and with even-order crossovers (i.e. 12 and 24 dB/octave), there is actually a phase reversal between the low frequency and the mid+high frequency outputs. This can be seen if one examines the wiring of a speaker using an even-order crossover network, and it will be observed that the midrange driver is wired out-of-phase with the woofer. The same thing happens with the mid to high crossover, except that the tweeter is now back in phase with the low frequency driver.

It must be noted that the phase reversal is required only to ensure that the drivers are in phase at the crossover frequency. A couple of octaves each side, and with the inductive and capacitive load presented by a loudspeaker, the signals are out of phase to a greater or lesser degree. This can only be considered a completely unacceptable situation when you look at it, but there it is - alive and well in a great many speaker systems from the very beginnings of high fidelity right up to the present day.

See [Project 09](http://sound.westhost.com/project09.htm) for a Linkwitz-Riley aligned 24dB/octave crossover that *is* phase coherent, and has the added benefit that there is exactly equal power at all times from the two drivers. A conventional crossover by way of comparison has a 3dB peak at the crossover frequency when the two outputs are summed.

I have made some amendments here, as I was previously caught out by "prior knowledge syndrome", a phenomenon that occurs when one has heard (and read) the "facts" so many times that one fails to check them properly. As it turns out, the so-called facts were wrong, and thus, so was I. With a resistive load, the phase response is as shown below, and the signals are 180 degrees out of phase at all times - not just at the crossover frequency. The problems arise because of the reactive nature of loudspeakers, which will mangle the phase response of almost any passive crossover. Many designs use complex networks in series and/or parallel with the drivers to attempt to get a net resistive load. This is not too difficult with the mid to high crossover, but is virtually impossible with the bass to mid+high crossover because of the very large resonances.

Figure 4 shows how the phase reversal in a 12dB/octave crossover comes about. At the crossover frequency, each waveform is subjected to a phase shift of 90 degrees. Since one is positive (called "leading" phase) and the other negative (lagging), the net result is that the two waveforms are 180 degrees apart - exactly out of phase. Notice that at frequencies significantly lower than the low-pass filter's cut-off frequency (defined as the -3dB frequency), there is little phase shift at all. The converse applies to the high-pass filter, so at significantly higher frequencies there is again little phase shift.

This gives rise to the phenomenon described above, where the driver phase reversal is needed to prevent massive cancellations at the crossover frequency. The cancellations will occur at other frequencies too, but are not audible because the level difference is so great.


Figure 4 - Frequency And Phase Response of 12dB/Octave Crossover

It has been demonstrated by many workers in the field of acoustics that absolute phase is inaudible. Indeed, if this were not the case, then moving one's head 300mm closer or farther away from a sound source would give rise to a massive change in the perceived sound. As we all know, this is not the case. It has also been shown that some waveforms sound different if the phase is reversed, but the definitive word here is 'different' - there is no right or wrong involved. This topic is dealt with in greater detail elsewhere on the ESP site.

However, we are not talking about absolute phase but relative phase - the situation where the phase of a signal is radiated from two different sources - *each with a different phase relationship from the other!* For this reason, many speaker manufacturers attempt to "time-align" the drivers so that the radiated signals are in the same physical plane - the idea being to combat additional phase distortion created by the loudspeaker drivers themselves. As the above shows, this is something of a lost battle - the crossover has already done plenty of damage to the phase response.

The only crossover which is relatively immune from the rapid phase transition around the crossover point as described above is a first order (6dB/octave) network, which is regrettably generally unsuitable for most loudspeakers because too much power is applied to the drivers outside their operating range. This can add considerable intermodulation distortion (this time loudspeaker induced), and is rarely an option in any system, especially between the low and mid+high frequencies. It may be an option with careful driver selection, but mostly this will only be applicable to the midrange to high frequency crossover (see below).

Although easily and cheaply built as an electronic filter, a passive third-order crossover is complex and expensive, and is more sensitive to variations in load impedance than the second-order filter.

No calculation is needed to demonstrate that if a speaker is pushed hard, its impedance will change - and this is completely aside from the reactive load presented. Most voice coils are wound using copper, and like all metals, copper has a positive coefficient of resistance. When you look at the specifications for most quality drivers, they will boast that they use a high temperature voice coil former - a good idea, since the voice coil can easily reach 150 degrees (Celsius) or more. This temperature change must cause a change in resistance, and any change will have an adverse effect on the alignment of the crossover, since impedance will change too.

Copper has a thermal coefficient of resistance such that its resistance increases by 0.39% per degree Celsius. Given a typical 6.6 ohm (DC) voice coil for an 8 ohm nominal speaker, at 150°C, the resistance will rise to over 10 ohms - naturally the impedance must be greater than this figure, so the loading on the crossover network is radically different from the design figure of 8 ohms.

At this point, the characteristics of the crossover are so far outside the design boundaries that any further calculation is futile.

1.12 - Phase-Coherent Electronic Crossover

It is possible to design a phase-coherent crossover (electronic, naturally), which exhibits none of the problem characteristics of the passive types. In an ideal world, the residual output (i.e. high frequencies below or low frequencies above the crossover point) would be in phase with the main output at any frequency in the spectrum. This ensures that there will be no cancellations or reinforcement of the signal as the outputs of the loudspeaker drivers re-combine in front of the cabinet. Regrettably, this is not the case with any passive crossover, and not all electronic crossovers exhibit this desirable characteristic either.

The author has designed and built such a unit however, and its phase performance is exemplary. All output signals remain absolutely in phase for two octaves or more above and below the crossover frequencies. With a correctly time-aligned enclosure, phase and transient response are quite extraordinary.

It should be noted that the original phase-coherent crossover built by the author is now nearly 20 years old, and resides in a loudspeaker test amplifier (mono, tri-amped, with sweepable crossover frequencies, variable impedance - the lot) and is still in regular use.

See [Project 09](http://sound.westhost.com/project09.htm) for a Linkwitz-Riley aligned 24dB/octave crossover that *is* phase coherent. The question of phase coherency seems to have come of age, as it were. I have seen several designs advertised that are phase coherent, and more speaker designers are striving to achieve this goal. This is as it should be, and I am most pleased to see this happening at last. Although the filter whose graphs are shown in Fig 4 is phase coherent, it requires a polarity reversal to ensure correct phase response.

Most electronic crossover networks will be phase coherent. The same cannot be said for passive networks, where amplitude and phase response is often dictated by the variable impedance of the loudspeaker drivers. While networks can correct these errors to a degree, they are not always used, and are not always sufficient to make a full correction.

Advertisers, reviewers, manufacturers and the listening public do not seem to have seen the benefits of a phase-coherent system, and most available models of electronic crossovers seem to be electronic versions of ordinary passive crossovers. Although these units still provide many of the advantages listed in this article, phase-coherence is not necessarily included unless the manufacturer specifically states that this is a feature of the design.

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| +  | Phase Response - A big gain for the bi-amped system, since it can be driven from a phase-coherent crossover eliminating the rapid phase variations around the crossover frequency, and no phase reversals between drivers. Transients are cleaner and the sound is more open than can ever be achieved using passive crossovers. |

1.13 - Bi-Wiring Facts And Myths (and More on Passive Crossovers)

Many speaker systems now cater for bi-wiring - running a separate speaker lead from the amplifier to the low and mid+high crossovers via separate terminals on the back of the enclosure. The benefits of this technique are said to be improved imaging due to the reduced interactions of the loudspeakers and their respective crossover networks, since the amplifier acts as an essentially zero impedance source for each section (the speaker cable now has no influence on crossover performance).

Some people equate bi-wiring as a cheaper method of achieving the same gains as one would with bi-amping. This is quite obviously not the case - there are gains to be had, but they are comparatively minor. This is not to say that the "minor" gains are not worth the effort, because as you will see this is not true at all.

For those who may not be sure of how bi-wiring really works, Figure 5 shows the setup. The broken line indicates where the connection would normally be made internally (i.e. inside the speaker cabinet). When bi-wiring is used, this connection is removed (usually with links on the connection panel), and a separate cable is run back to the amplifier.

This diagram also illustrates the composition of a 3-way crossover network. Low frequencies are fed to the woofer via a low-pass filter. The remaining signal is then fed through a high-pass filter to remove the bass energy. This is the mid+high component. To ensure that the midrange loudspeaker does not receive high frequencies as well (which it would otherwise proceed to mangle), a low-pass filter is used to filter out the high frequency component. Finally, to ensure that the tweeter is protected from the midrange signals, another high-pass filter is used.

The cutoff (i.e. crossover) frequencies for the two filter "groups" will typically be in the range of 300Hz to 800Hz for the low/mid+high section, and 2kHz to 6kHz for the mid/high section. Crossover frequency selection is discussed in a following section. ([A Short Dissertation On Crossover Frequency Selection](http://sound.westhost.com/bi-amp2.htm#freq_sel))


Figure 5 - Bi-Wiring Connections

As can be seen, the low frequency energy is now separated from the mid+high frequency energy in the cabling. The amplifier must still handle the full frequency range, but each section of the crossover has its own cable feed, which prevents some of the interactions between the crossover sections.

The overall effect is often (or so it has been said) a vast improvement, and largely because of the fundamental imperfections of passive crossover networks. These networks (regardless of their cost or complexity) have a few basic weaknesses which determine their overall performance. Basically, these are:

Input impedance - a properly designed inductor / capacitor (L/C) filter relies on the input impedance (i.e. the impedance presented by the amplifier and speaker leads) to be zero ohms if the design crossover frequency is to be achieved. Any variation from this ideal will cause a small but measurable shift in the crossover's operating characteristics. When two such filters (low pass for the woofer and high pass for mid+high) are combined and share a common resistance (speaker leads) back to the source (the amplifier), interaction is inevitable, with the resulting final frequency and phase variations being unpredictable in real life.

Load impedance - for any passive crossover to achieve its design frequency and phase response, the load must be purely resistive or wide variations in response are unavoidable. To combat this, many manufacturers design often highly complex impedance correction networks, placed across the loudspeaker in an attempt to make it "appear" resistive to the preceding filter. This task is by no means simple - a loudspeaker driver is a coil of wire (having resistance and inductance), connected to a moving mass (the diaphragm, cone, dome or whatever), all enclosed in a housing with its own resonances and reflections and coupled to the outside world via the air. This represents a highly complex electro- mechanical circuit, and presenting a pure resistance to the crossover can be approached, but never achieved (at least not over the desirable minimum of two octaves).

In addition, these impedance correction circuits will invariably introduce additional losses, since they will generally consist of a combination of inductors, capacitors and resistors to achieve the desired result. Indeed, these networks can become more complex than the crossover itself in extreme cases, adding significantly to the final cost of the speaker system.

Internal Impedance - as previously discussed, the coils (inductors) used in a crossover network must have resistance, since they are simply (hopefully!) an air-cored coil of copper wire. Being air-cored is the ideal, with no ferrous materials used at all. Using an iron (or any other magnetic material) core increases the inductance of the coil, so for the same value of inductance it will require fewer turns. Although this reduces internal resistance, it is at the expense of linearity and distortion, both of which are adversely affected by the use of magnetic materials.

So, a good quality crossover inductor will have many turns of wire, which naturally has resistance. As discussed above, this should only be reduced by using a heavier gauge of wire - but compared to the cross-sectional area of your speaker cables, it is still going to be very thin! Again, power loss and variations from the ideal crossover frequency are the direct result. Naturally, speaker manufacturers are going to make the necessary frequency corrections for the introduced resistance (in reality this is very small), but can do nothing to eliminate the power losses due to resistance. The crossover resistance also affects the damping factor presented to the loudspeaker by the amplifier - adversely of course.

Variation of the Mechanical Components - The materials used for the manufacture of loudspeaker drivers are "real-world", which is to say they are imperfect. The roll surround of a woofer will not present the same impedance at all temperatures, and will change with age - as will all the other components. Of the fundamental ingredients which determine the overall characteristics of a loudspeaker, only the air is predictable over time (with the effects of greenhouse gasses, even this is debatable). But as we know, air varies its density and mass depending on altitude, barometric pressure, temperature and humidity, and even these variations will have some small effect on the overall characteristics of the driver. Since the response of the crossover network is dependent on all these mechanical factors as well as the electrical characteristics, it is small wonder that speaker systems seem to change their sound over time.

Of the components which change over time, the loudspeaker itself is the greatest offender. Roll surrounds will stiffen (or start to decompose and "relax"), the cone will lose some of its stiffness and the spider (the section of the speaker which supports the voice coil) will also change. Damping materials in the speaker cabinet may settle or become more compacted as well, again changing the load presented to the crossover filter.

Thermal Effects - So far, this discussion has taken into account only briefly the fact that the loudspeaker voice coil and inductance coil(s) in the crossover are generally of copper, and that its resistance will change with temperature. Particularly at high power levels, the self heating of these components cannot be neglected, so as the temperature increases, so too do the losses. There is also the small matter of the variation of load impedance presented to the crossover network as the loudspeaker voice coil heats up. Remember that the passive network relies on the load resistance (pure resistance is essential) remaining exactly the same at all times. Although these effects are measurable, it has neither been proven nor disproved that they are audible. I know of no speaker manufacturer who has included thermistors (thermal resistors) in a crossover network to maintain the exact design resistance at all voice coil temperatures. Indeed, the use of these devices would probably degrade the sound, since they are affected by self-heating (by the current flowing through them) which will vary with the signal.

The use of oxygen free copper 'super' cable for speakers has many audiophiles convinced of the huge improvement (an argument I will not be drawn upon in this article, except for the note [[1]](http://sound.westhost.com/bi-amp.htm#note1) below) - yet the variations in frequency response caused by the inductance, resistance and capacitance of the cable are barely perceptible (if at all) with the most sophisticated test equipment (other than ears - supposedly). If these barely measurable effects are so apparent to listeners, why would the response variations of a crossover caused by temperature not be detectable? I believe that these changes are detectable, but since all (passive) speakers suffer from the same problem, this has simply been accepted as normal. These changes are also gradual, which allows time for our ears to adjust and compensate - this is now delving into the area of psycho-acoustics, which I may examine in a future article - but not here and not now.

Again (this is starting to get tedious), an electronic crossover and separate power amplifiers will virtually eliminate the thermal effects other that a slight reduction of power as the loudspeaker voice coil heats. Even this can be corrected but the effort is not worth the gain, and due to additional circuit complexity will probably subtract more than it adds!

Many people have said that bi-wiring improved the sound quality, and although I have not used it myself (bi-amping being so far superior), I will reserve judgement until further notice. While there are some measurable differences, if sensibly sized cables are used the difference is unlikely to be audible unless the loudspeaker's crossover network has serious anomalies.

For more detailed information on the design of passive crossovers, and the many pitfalls involved, read the article [Passive Crossover Design](http://sound.westhost.com/lr-passive.htm). While a mid to high passive crossover can be made reasonably economically and if well designed can sound very good (even excellent), there is often much to be gained by using a fully active system, where each driver has its own amplifier.

Note 1: It must be said that speaker cables in reality contribute little in the way of problems in hi-fi equipment. Much has been made of 'super' cables and the like, but in reality although measurable at audio frequencies, there is no proof that these effects are audible to the majority of listeners. The levels of performance variation caused by the cables are in fractions of a dB, so provided a sufficiently sized cable is used, it matters not whether it cost $2.50 / metre or ten times that amount - except to one's bank account, of course.

Note that the above may not necessarily be the case ... most speaker cables are benign, but some can easily cause a marginal amplifier to oscillate. In particular, be wary of those marketed as having a characteristic impedance of 8 ohms or others offering very low inductance. For more information on this topic, see [Cable Impedance](http://sound.westhost.com/cable-z.htm). There are solutions for these pointless (IMHO) constructions, but it's best to avoid them altogether. Any cable that is capable of making an amplifier unstable cannot offer an improvement, but can easily cause potentially serious problems.

To summarise this section, the complexity of a well designed passive crossover will be such that it will add significantly to the price of the speaker system, while still resolutely presenting the same old problems - power loss, interaction, imperfect impedance matching and rapid phase shifts around the crossover frequency. Variations of loudspeaker driver performance caused by ambient conditions (temperature and humidity for example) will affect all speaker systems (including bi-amped), but will only cause crossover frequency shifts with a passive network.

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| +  | So now we can add a couple more plusses for bi-amplifying one's system - complete freedom from any interaction between the loudspeaker driver (and its environment) and the crossover network, and a potentially large cost saving for the now redundant complex passive crossover network.Add to this the benefit that many listeners already experience from bi-wiring - since a bi-amped system must be naturally be bi-wired since this is a fundamental part of its operation! |

1.14 - Midrange to High Frequency Crossovers

So far I have only described the low to mid+high frequency crossover, because this is traditionally the most complex and power-hungry part of the network. There are potential gains to be had by tri-amping a system, but they become hard to justify when the added cost and complexity are considered.

The midrange to high frequency crossover still suffers from the same ailments as the low to mid+high, but they tend to be overshadowed by response aberrations created by the edges of the enclosure, grille cloth frames and other discontinuities. For this very reason, many manufacturers now use felt pads around the tweeter to reduce these effects. To some extent, the loudspeakers themselves will generally not be quite as difficult at higher frequencies, because the effects of cabinet resonances, cone mass and other mechanical factors are not as severe as at the low frequency end of the spectrum. This is admittedly a simplification, but the effects are more subtle unless the designer happens to be pushing loudspeakers beyond their limits.

Ferro-fluids are often introduced into the magnetic gap of mid and high frequency drivers, improving magnetic coupling and reducing resonances by way of the additional damping. This technique cannot be used on low frequency drivers, because the excursion of the voice coil is too great and the fluid would simply be flung from the gap as the cone moved.

Since the loudspeakers are actually more controlled at high frequencies, the complexity of the crossover is reduced. Some impedance correction is nearly always needed for the midrange driver (as its impedance will tend to rise with increasing frequency), but this is not especially arduous.

Likewise, padding may well be needed to make the driver sensitivities effectively equal, but especially in a bi-amped system this should only ever be done to reduce the sensitivity of a tweeter to match the midrange - never the other way around.

If one were to obtain drivers whose frequency response and power handling allowed it, a first-order 6dB/octave crossover network is ideal - good transient response (the best of all filter types, in fact), freedom from phase aberrations, no polarity reversal of drivers - the list goes on.

1.15 - Summary

During this article, I have given great account of the benefits of bi-amping, but nary a word about any negatives in the equation. There are some, of course, but they have actually been described already ...

Two power amplifiers are required for each channel of a stereo system

An electronic crossover is needed

An ideal solution would be to incorporate the power amplifiers and electronic crossover within the speaker cabinets themselves. This has been done by many manufacturers (including the author, many years ago), and is becoming quite common for high-end studio monitor speakers, especially some of the new "near-field" systems.

Even using conventional amplifiers and a separate electronic crossover network, real advantages are to be had. Imagine the best of all worlds - a really good transistor amp for the low end, providing tight and well controlled bass, and a valve (tube) amplifier for the midrange and high frequencies. With its soft overload characteristics and the renowned openness of a good valve amp, with none of the standard valve amp failings - woolly bass (generally due to the rather poor damping factor of valve amplifiers), and low frequency intermodulation distortion, caused primarily by the output transformer.

|  |  |
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| +  | Another great benefit has revealed itself. Complete flexibility to choose amplifiers which are at their very best within a defined frequency range. Now the amp which all the reviewers said has "magnificent bass - but is disappointingly lacking at the top end", and the other one which is "glorious at the higher frequencies but suffers from lack of bass extension and tends towards woolliness" at low frequencies can find homes where they really excel! |

1.16 - Adding up the Plusses

Effectively (up to) twice the 'real' power of the amplifiers themselves

Reduced intermodulation distortion

Elimination of the low frequency passive crossover, its inherent losses, potentially poor linearity and crossover point inaccuracy

Reduction of the difficulty of the load presented to the power amplifier

No padding is required to align the driver sensitivities, so we are not simply wasting power

The damping factor is greatly improved for both the low and midrange loudspeakers

Complete freedom from any interaction between the loudspeaker driver (and its environment) and the crossover network

Cost savings, since complex passive crossover networks are not needed

Bi-wiring is included free!

The flexibility to choose amplifiers which are at their best within a defined frequency range

Ability to match amplifier power to the exact requirements of the drivers for maximum overall efficiency

I could go on (and on) here, but I shall resist the temptation. There is (IMHO) no reason to not use biamping wherever possible, from small (i.e. computer) speakers through to top of the line hi-fi. The benefits far outweigh the disadvantages in all cases.

I have seen many claims that loudspeaker manufacturers often go to extraordinary lengths to design the best possible crossover network for their products. I do not doubt that for many high-end systems, this is certainly the case. It must also be considered how much extra this costs, and we can be assured that many systems have a less than ideal network, simply to keep costs reasonable. Several times, I have seen reviews where expensive speakers use ferrite cored inductors for the low frequencies, and bipolar electrolytics are also common.

I do not consider these to be optimal or appropriate for a high quality system, and nor do many others. The truth is that cost considerations are nearly always made in any system, and much more so when the selling price becomes a consideration.

As I stated at the beginning, if you spend $25,000 or more for a pair of speakers, then we are into the "cost no object" area. Most people cannot afford such luxuries, and as a result they must settle for something they can afford. Only a very few systems will be as good as they can be, and you will pay dearly for it.

Biamping is not a simple tweak, and is not to be taken lightly. Make no mistake though, its application *will* improve almost any loudspeaker available, with very few exceptions.

There is one area where you may find that changing to a biamped system cases an apparent loss of bass response. This is rare, but in some cases the cabinet design may have been optimised for the woofer, including the resistance of the series inductor in the passive crossover. This is especially true for very well designed ported boxes. The result is that a biamped system is slightly over-damped by comparison, resulting in a loss of bass response.

There are two solutions for this ... either allow the system to 'break in' (which actually means that you get used to the new sound after a while), or increase the output impedance of the power amp. See the ESP articles index for more information on how to go about this. In general, the impedance variation is small, but it is certainly worth doing if you wish to get the best possible response from the system.

2.0 - Crossover Frequency Selection (A Short Dissertation)

It would be remiss of me to not mention a few salient points about the choice of crossover frequencies. This applies to all system types where fidelity is expected (or demanded) - high power music, sound reinforcement, or hi-fi.

It is not at all uncommon to see systems where the crossover frequency is set right in the middle of what I call the 'intelligence band'. This is the range of frequencies from 300Hz to 3600Hz, and is extremely important from a psycho-acoustic point of view.

It is no accident that this is the range of the telephone system, and has been for many years - ever since electronics became involve in telephony. If we are only to hear a limited range, then this band of frequencies is by far the most important. Just from this we can recognise a person's voice, which musical instrument is being played (even bass instruments!), and - more importantly - what is being said. It contains nearly all the 'intelligence' of the sound, which is to say that if this band is corrupted, intelligibility is greatly reduced.

So why do speaker manufacturers insist on placing their crossover frequencies within this band of frequencies? The public address (PA) systems used by many rock bands are a case in point - how often does one find that the vocals are completely unintelligible? Mind you, it may also be the case that the band's lyrics just don't make sense, but that's another story altogether.

Often this occurs because the system is so loud that the amplifiers are clipping badly, but even at lower levels it is quite common. Place a common-or-garden crossover filter right in the middle of the "intelligence band" and this is exactly what will (and does) happen. With phase aberrations and cancellations, this most important frequency range becomes muddied and indistinct causing loss of intelligibility - not only on voice, but instruments as well.

The effect is also noticeable with some hi-fi speaker systems, except that it usually less pronounced, and it is far less likely that the amplifier will be driven to clipping. Reviewers will often say of a speaker that the vocals seem veiled, or that there is noticeable colouration of either male or female vocals. These effects are often caused by the effects of phase shift around the crossover frequency, coupled with the fact that the crossover frequency falls right in the middle of the intelligence band.

Should a crossover be unavoidable in this region - due (for example) to available loudspeaker drivers - then the manufacturer must go to great lengths to ensure that artefacts created by the crossover are not audible. This often causes greater problems with amplifier loading at the crossover frequency, since impedance dips seem to be a common problem with many speakers. It will be found that these almost invariably occur at the crossover frequency.

By using an active crossover network, it should be possible to get excellent performance almost regardless of what the crossover frequency may be. The final setup will still have to be carefully aligned to make sure that there are no major issues with either driver at the selected frequency. In the course of many experiments and tests, it is safe to say that a properly set up active crossover gives one far more flexibility than almost any passive version, with the great advantage that no loudspeaker impedance correction is needed.

2.1 - Ideal Crossover Frequencies

Since we have already discussed the 'equal power' crossover frequency between low and mid+high frequencies, it should come as no surprise that the author prefers between 275Hz and 300Hz as the ideal frequency. This is outside the intelligence band (albeit only just), but as discussed, a phase-coherent crossover network and a bi-amped system will tend to be far more tolerant than conventional (passive) crossover networks.

One problem this technique does cause, is that the demands placed on the midrange driver are greater than will normally be the case. This is because the low frequency end of the midrange is now extended to around 300Hz rather than the more 'conventional' frequency of 500 or 600Hz. Few (none that I know of) so-called 'enclosed' (i.e. those with their own integral enclosure) midrange drivers are capable of reproducing 300Hz accurately - indeed, many are quite inadequate even at 600Hz!

Even ruling out this style of driver altogether still leaves relatively few speakers which are small enough to be considered a point source at 3kHz (one wavelength at this frequency is only 115mm - assuming 'British Standard' air temperature, etc.), yet is capable of reproducing signals down to 300Hz accurately. Ideally one would want a driver whose radiating surface is no greater than 100mm diameter (this is already a significant compromise), having high compliance for low frequency reproduction, and a stiff cone structure to prevent cone break-up at the upper limits.

Bear in mind that a loudspeaker which is going to be used to reproduce frequencies down to 300Hz should ideally be capable of uncoloured reproduction for at least one octave (and preferably two octaves) above and below the crossover frequencies. This means that a suitable midrange driver must be capable of reproducing from 150Hz to 6kHz with good efficiency and without significant colouration. This is not an easy task for any loudspeaker.

Many otherwise fine midrange drivers do not provide a wide enough safety margin below their recommended minimum crossover frequency, which causes resonances and other effects to colour the sound. Also affected will be phase response, which will start to suffer badly as the driver approaches resonance - this rather negates the advantages of using a phase-coherent crossover network!

2.2 - The Ideal Compromise

Yes, I know the heading is an oxymoron, but that is what we really have to find. We cannot go further into discussion at this point (at least not without naming names, and deciding on some suitable loudspeaker drivers), since the 'ideal compromise' will be different for every loudspeaker combination available, with added problems incurred by the selected cabinet design and the maker's design goals (price - as always - being a major player in all these calculations).

Having examined some of the factors which affect the performance of a speaker system, it is apparent that there are few hard and fast rules which can be applied, since there are so many variables. What has been presented here is a guideline which - assuming that suitable drivers can be obtained - will have a standard of performance well above average. This web site has now been updated many times, as more information comes to hand, and as I get responses from readers who have similar (or wildly different - rare!) views from my own.

It is to be hoped that this information will at least provide some further discussion and feedback from readers who share my interest in "the ultimate loudspeaker" - however it is configured (even with passive crossovers, perhaps).

2.3 - 2-Way Systems

Not everyone wants to use a 3-way or 4-way system, and just want to use a mid-bass driver with a tweeter. Although it's often difficult to keep intermodulation distortion down to respectable limits with such systems, at moderate listening volume they are often all that's needed for smaller rooms. They are a popular choice, and there are many very good kit designs on the Net and in magazines. Indeed, I have a pair of 2-way boxes in a back room of my house, and that's all that is needed for casual listening to music or while watching TV. Mine are still passive, but active 2-way systems are much better in almost all respects.

When you go to 2-way active, don't expect to get a useful increase in SPL or effective amplifier power - when you cross over at around 3kHz there is almost nothing to be gained there. What you *do* get is a crossover that behaves itself properly, and you no longer need to add Zobel networks or other impedance correction techniques. You also get the ability to play with different amplifiers for the two sections. Although most of the differences you are likely to hear are likely due to the 'experimenter expectancy effect', if you do happen on a combination that you find especially pleasing, then you still win.

In general, the mid-bass driver for any 2-way system will need to be in a vented enclosure to minimise cone excursion at low frequencies. A sealed box demands far too much voicecoil travel, and few drivers will remain linear if pushed anywhere near their claimed maximum excursion. As with all loudspeaker designs, this is yet another example of the compromises that must be made to achieve a result that you'll be happy with.

For most systems of this type, there is a major compromise between mid-bass diameter and crossover frequency. Because few tweeters will tolerate frequencies much below 3kHz, the upper frequency of the mid-bass is pushed to its limits. Off-axis response is usually very poor, because the driver diameter is commonly greater than the wavelength at 3kHz (for example). When piston diameter approaches the wavelength of frequencies to be reproduced, the speaker starts to 'beam' - there is a major lobe directly in front of the driver, and lesser lobes (which change with frequency) as you move off-axis. Some driver datasheets include a polar plot showing the off-axis response lobes and nulls, while others just show the frequency response on and off-axis (typically 45°).

The mid-bass driver's upper limit and the tweeter's lower limit determine the crossover frequency that you have to use. You can use a waveguide to allow the tweeter to operate down to a lower than normal frequency - see [Practical DIY Waveguides](http://sound.westhost.com/articles/waveguides1.htm) for more information on this technique. Using a 24dB/octave electronic crossover (such as ESP's [Project 09](http://sound.westhost.com/project09.htm), you can sometimes run the tweeter to a lower frequency than that recommended, because the default crossover network is nearly always a 12dB/octave passive design.

By using an electronic crossover with sharp cutoff, there can be less stress on the tweeter than normal, even if the frequency is reduced below that recommended by the manufacturer. You need to be very careful though, because tweeters are easy to damage with lower than normal/ recommended xover frequencies. Making changes needs a good understanding of the possible ramifications, which include but are not limited to tweeter failure. In particular, make sure that the available power that can reach the tweeter is limited - use of a 100W amp (for example) is a very bad idea indeed.

The following section discusses the other issues that you need to address. From the tweeter's perspective, it doesn't matter if the midrange driver is dedicated to midrange, or handles everything from the tweeter crossover frequency down to perhaps 40Hz or so.

3.0 - Tri-Amping

I have had many enquiries about extending the bi-amp principle to tri-amping, and offer a few thoughts here. There are some points which must be made, largely to protect the tweeters in such a system, but also to ensure that the system as a whole is coherent, with no one component of the music receiving more or less attention than the others.

Three-Way speaker systems offer many advantages, and the extra cost of making the whole system active is comparatively small. The increase in performance will depend on how good (or not) the passive crossover section might be. 'Good' means that the system uses a high quality passive section between the midrange and tweeter, and includes impedance correction, *and* has no bad habits from the amplifier's perspective. Bad habits include impedance dips - some speakers may have the impedance falling to less than 2 ohms at the crossover frequency for example.

Even if the passive crossover is really well designed, it is unlikely that it will beat the performance of an active system. The reasons are quite simple - with an active system, each loudspeaker driver has its own amplifier (operating over a limited frequency range), and there is zero mutual electrical interference between the drivers. No passive crossover can achieve this, because even a small impedance variation from one driver affects the performance of the other. Inductors are imperfect (to put it mildly), and it is economically unrealistic to attempt to get any passive crossover to be the equal of its active counterpart.

However, there are still things that one must be aware of, as discussed below ...

3.1 - DC Protection

With a bi-amped 3-way system, the tweeters are protected by the mid-high passive crossover. Once the loudspeaker is tri-amped, this protection is lost, since the capacitor which is used to determine the crossover frequency is no longer present.

With most 'solid-state' amps, this places the tweeter at great risk during the (generally short) switch-on and switch-off periods. As the supply voltage is applied (or removed), some amplifiers will create a DC transient (if such a thing is possible) as the circuitry starts to operate. This causes the all-too-common speaker thump.

This is mildly annoying when applied to the low frequency drivers, but is capable of destroying a tweeter if allowed to persist for more than a few milliseconds.

In the case of amplifier failure, the tweeter is almost certain to protect any speaker fuse by blowing first - not exactly the desired effect! The 'Poly-Switches' now available might help, but I don't like the idea of a non-linear resistor in series with my speakers. Having said that, Poly-Switches are certainly a viable way to protect a tweeter, but not from short-term DC. They are fairly slow-acting, and are more useful for providing protection from long term overpowering ... such as an amplifier driven to clipping for example.

If the direct coupled approach is contemplated, I would suggest the following:

Use a relay in series between the tweeter and the amp with a suitable muting circuit

Include a fast acting DC sensor to disconnect the load if amplifier DC is detected

A suitable circuit is available - see [Project 33](http://sound.westhost.com/project33.htm) in my Project Pages, which can be easily be modified to protect tweeters, where its DC detection circuit can be made very fast indeed.

3.2 - Choice of Capacitor

A humble capacitor will prevent DC from reaching the tweeter voice coil, but the selection is critical to ensure that the sound is not degraded.

Value - The capacitor will almost always have to be at least 20uF, which for an 8 Ohm tweeter, will create a 3dB high pass crossover at about 995Hz. Given that this additional crossover should be ideally 1.5 to 2 octaves from the 'real' crossover frequency (even more if possible), the values likely to be needed in real life will tend to get quite large. The reason that the protection cap needs to be so large is that smaller values introduce phase shift, which is significant for all frequencies within 2 octaves of the crossover point.

An alternative (I hope your maths are good) is to use a modified high pass section in the electronic crossover, and then use the protection cap to provide the last pole of the filter. This will work (it will work very well), but the mathematical complexities will be such that I expect few constructors to go this way.

A further disadvantage is that the electronic crossover cannot simply be swapped for a different type to allow comparisons, and with some filter types the approach will not work at all.

Type - When we contemplate high value caps (greater than 20uF) there is an immediate tendency to think about using a bipolar electrolytic. For this application, I do not recommend them, but sometimes you may have little choice. According to some, they are not recommend for any application, since they are (supposedly) sonically disgusting. I have not been able to measure distortion in a bipolar electro, but there are many who claim that they destroy the sound. I shall not continue this debate.

The ideal is to use polyester or polypropylene caps, since their stability is so vastly superior to bipolar electrolytics that there is no comparison. They also have a comparatively unlimited life, but bipolar electrolytics gradually lose capacitance (and sometimes not so gradually), thus changing the crossover frequency (or disabling the tweeters completely when they eventually fail.

Good caps can cause some degree of financial hardship, but be assured, that is as nothing compared to the utter despair when smoke is seen escaping from your precious tweets.

If you are on a budget (decent caps at these values are expensive), one possibility is to use power-factor correction or induction motor start capacitors. These used to be oil-filled paper (some still are), and are much cheaper than 'electronics shop' devices. I can vouch for the sound quality, as I use these to protect my tweeters - most are polypropylene are of film and foil construction, although metallised film is probably used as well. The stability and power handling will certainly be superior to that of bipolar electrolytics, and the high frequency response can be corrected (if necessary) by bypassing with a high grade polyester. I would expect that a 1uF bypass would suffice in most cases.

These caps should normally be available from electrical supply outlets, since they are commonly used in electrical (i.e. mains house/ factory/ office) installations.

3.3 - Amplifier

The amplifiers for a triamped system may have an effect on the final sound quality. This is especially true of the tweeter amp, which will generally not require a lot of power (depending on crossover frequency). If we assume that the power drops off at 3dB/octave above 1kHz for 'typical' music signals, we can do a quick calculation - this is not difficult (nor is it terribly accurate), but will give an idea of how much power will be needed for the tweeters. Note that this formula errs on the side of safety (i.e. the tweeter amp will have more power than is really needed), and this provides a good margin - a tweeter driver amplifier which is clipping is not likely to enhance the sound quality !

We might quickly re-examine the power of the low and mid amps first, assuming that we have selected the 'equal power' low/mid frequency of about 300Hz. For a typical system for home use, 50 Watts for each will generally be enough - especially when you remember that biamping can give up to the approximate equivalent of double the actual power of the amps - i.e. 200 Watts.

So, for this example, given that we have arrived at using a 50 Watt amp for mid+high, we are now going to triamp, with a crossover frequency of (say) 3kHz. This is approximately 1.7 octaves above 1kHz (it's a little more, but it is not worth worrying about).

At 3dB/octave, and 1.7 octaves, this results in a power requirement for the tweeters of -3 x 1.7 = -5.1dB relative to the midrange amplifier. Reversing the dB (power) formula, it can be seen that the high frequency amp will need 0.31 of the midrange amp's power.

0.31 x 50 Watts = 15.5 Watts. I suggest that a 20-25 Watt amp will be appropriate, and will have more than enough headroom. This hypothesis has been proven in practice - my own system uses 70W midrange amps and a 20W tweeter amp. I doubt that it has *ever* clipped since the system was first set up.

3.4 - Class-A Amplifiers

For 20 Watts, we can look seriously at using a Class-A amplifier, something that most of us would die for, but is unrealistic for higher powers.

High power Class-A amps are seriously expensive to build or buy, and create a lot of heat. At the small power of 20 Watts however, they start to become much more attractive. They still create a lot of heat, but since this is proportional to their output power it becomes manageable at low powers.

A typical 20 Watt Class-A amp will dissipate about 100 Watts worst case, and although this is not insignificant, it can still be dealt with by conventional heatsinks and no fan cooling. This is not to say that the heatsinks will be small - they most certainly will not - but 50W per device (assuming transistors) is not too hard to get rid of.

At these powers, one might even consider a valve (vacuum tube) Class-A design, but I would not be inclined to this approach (personal opinion), however it may be that this could make musical magic. It you try it and love it, then you have a winning combination. Bear in mind that some tweeters do not like being driven with any appreciable impedance - the response may become uneven, with lots of small deviations from the ideal. In such cases, you need to use a transistor amp - the output impedance of most valve amps is at least a couple of Ohms.

Schematic Diagram

Refer to the Project Pages for a design of a couple of transistor 15 to 20 Watt Class-A amplifiers designed for general use, but are ideal for driving tweeters. Includes the design and basic/generic construction details. As yet, I have not had time to test one of these circuits, so final specifications are not complete, but the DoZ is a fairly nice little amp.

Also, have a look at the (now old but still useful) design - [10 Watt Class-A Amplifier](http://sound.westhost.com/jll_hood.htm) (By John Linsley Hood). You can also have a look at [Project 72](http://sound.westhost.com/project72.htm). While the LM1875 or TDA2050 is not normally suggested for true hi-fi, this is by omission rather than due to any major deficiency (although the TDA3050 is a better choice). However, a TDA7293 or LM3886 based amp is probably seen as more appropriate, and it's hard to argue against this.

4.0 - Correcting Crossover Filter Amplitude Response

When an electronic crossover is used together with the respective amplifiers for each channel, there is always going to be a temptation to experiment with the levels of the filters or amplifiers to act as a sort of tone control. To extract the maximum benefit from bi- or tri-amping, this should not be done, since it will effectively do a few things (all undesirable)

The crossover frequency will be modified

The phase relationships between crossovers may no longer be valid

The frequency response will have peaks or dips (or both) where it should be nice and flat

The optimum settings for the relative gains are dependent on only one thing - speaker sensitivity.

In order for this to make some degree of sense, we must return to our previous examples, and look at a few more diagrams. First, let's look at the ideal, where both speakers have a sensitivity of 90dB/m/W. In this case, the gains of the crossover sections (if gain controls are provided) should be exactly equal. Likewise, the sensitivity of the power amplifiers must also be equal.

In many cases, different amplifiers will be used, often with differing power ratings as well. This is where some measurements are needed, since both amps must have the same gain.

4.1 - Measuring Amplifier Gain

This is quite easy to do, but you do need a single frequency stable tone source - music is of no use, because it is too dynamic so levels are constantly changing. However you might consider the use of a test CD, which will have various frequencies at predictable levels. In many respects, this will be easier to use than any other method, since it requires only that the CD is inserted, rather than dragging oscillators or other signal sources about.

Unless the amplifier is a valve unit, it is not necessary to have a speaker load connected for these tests, or a suitably high-powered resistor can be used as a load if you want.

If a known level (say 100mV) is injected into the input of the amp you are going to use for bass (for example), you will measure an output voltage of about 2.5V at the amplifier output (this represents a fairly typical gain of 28dB). This must be identical for the amp being used for mid+high - assuming that the speakers have the same sensitivity. If the gains are not the same, you must install a volume control on the amplifier whose gain is the higher, and adjust until both amps produce exactly the same voltage at their outputs for 100mV input. The ESP Project 09 crossover has trimpots to allow the levels to be set.

For the test frequency, use an oscillator at about 400Hz, or if you don't have an oscillator at all, you can use the attenuated output from a small power transformer. This will not be as good, but it will work. The frequency will be either 50Hz or 60Hz, depending on your local supply. (If you don't know how to use a transformer to do this, ask someone to help - you can easily damage the input stage of the amp (and the rest of it !) if the level is too high). Alternately, use a test CD as mentioned above.


Figure 6 - Test Setup For Gain Measurement

The dummy load resistor should be equal to the speaker nominal impedance, and be rated at 5 to 10 Watts. Do not attempt to operate the amp at full power (especially if rated at more than 20W) into the load, or it will get very hot indeed. If you want to do this, then the resistor power rating should be at least double the expected amplifier output power. (Either that, or use lower power resistors and suspend them in a bucket of water - it will not cause a short circuit, fear not).

The voltmeter used may be digital or analogue - as long as it can read the voltage at the test frequency accurately - note that not all can do so!

4.2 - Speakers With Different Sensitivities

This is where things start to get a bit tricky. You will need to be able to calculate the required gain to suit the speaker sensitivities - not hard, but you might find that the scientific mode on the Windows calculator is useful (unless you already have a full scientific calc, of course).

Depending upon the crossover frequency, you might need to use a higher powered amp for the bass end, if its speaker has a lower sensitivity. For the purposes of the exercise, we will assume that the midrange (plus high frequency) speakers have a sensitivity of 90dB / W @1m as before. But the woofers have a sensitivity of 88dB / W @1m so we need to calculate the power and gain differences (assuming that the 'equal power distribution' frequency of about 300Hz is being used - you want to use a different frequency? If you follow these procedures, you will become an expert at this stuff - guaranteed - because you will have to determine the relative power levels for the crossover frequency you are using - and I'm not going to help !

First, we will calculate the gain difference. Assume that the mid+high amp has a gain of 28dB, so the bass amp needs a gain of 30dB (the speaker is 2dB less efficient, so we just add the 2dB to the 28dB of the mid+high amp).

We will use the same 100mV input signal, so:

Gain = antilog (dB / 20) = antilog (30 / 20) = antilog (1.5) = 31.623

Since we started with 100mV (0.1V), the output voltage must be 3.16V from the output of the amp. That wasn't so hard. Now we need to determine the power output of the bass amp, if it is to exactly match the mid+high amp. Let's assume that we will use a 50W amplifier for the mid+high (with 28dB of gain).

P2 = antilog (dB / 10) \* P1 (where P1 is the known power, and P2 is the unknown (higher) power)

P2 = antilog (2 / 10) \* 50 = antilog (0.2) \* 50 = 1.585 \* 50 = 79.25W

Note that with the power calculation, the value of 10 is used, rather than 20 for voltage or current calculations.

We have now discovered that an 80W amplifier is needed with a gain of 31.6 (30dB), to exactly match the amp power and speaker efficiency of the mid+high combination.

Ah. So you have measured the amps, and have an output voltage, but cannot relate him to decibels. Fear not, another formula is at your disposal:

dB = 20 \* log (V1 / V2)   (where V1 is the higher (i.e. output) voltage and V2 is the smaller)

So if you measure an output of 2.32V at the output of the amp with an input of 100mV, its gain is ...

dB = 20 \* log (2.32 / 0.1) = 20 \* log (23.2) = 20 \* 1.365 = 27.3dB

Note that in all calculations I have rounded the values to 3 decimal places, but when you do the calculations, retain all decimal places available for best accuracy. The difference is not great, but there is no need to introduce inaccuracies for no good reason.

4.3 - The Effect of Amplitude Inaccuracies

To see what happens when the gain is not correct, we need to look at the crossover curves again. Refer to Figure 7 - the red and green traces. This is the optimal frequency response of the crossover/amplifier/speaker combination, with the resulting output being virtually flat (there is a slight rise at the crossover frequency which can be corrected using Linkwitz Riley alignment, where the crossover point is 6dB down - see [Project 09](http://sound.westhost.com/project09.htm))

Have a look at what happens when the amplitude of one filter is different from what it should be. This is also shown in Figure 7, and it is clear that the effective crossover frequency is shifted. What is not so clear is the final frequency response, and in the case of any crossover filter that is not phase-coherent, the adverse effects of the relative phase relationships. These are extremely difficult to quantify, but may be apparent in listening tests. The problem is that if you are unaware of the problems that can be created by modifying gain indiscriminately, it will be very hard indeed to determine why the system just doesn't sound right.


Figure 7 - The Effect of Changing Gain on Crossover Frequency

It is very obvious that the effective crossover frequency has changed. At normal gain, the crossover is 295Hz, but if the gain is increased as shown, the crossover frequency shifts up to 500Hz. If the gain is reduced, the effective crossover frequency is now about 150Hz. Naturally, the same thing happens if we change the mid+high gain. Note that the filter cutoff frequencies are not changed, only the level with respect to the adjacent filter.

This is not just the output of the filter we are looking at, but rather the final output from the speakers - as shown above, it will often be necessary to change the gain of amplifiers to match the efficiency of the loudspeaker drivers used. This does not alter the crossover frequency as you might expect, but brings it back into proper alignment. In fact, if the gain is *not*changed, then you will get a result similar to that shown, by effectively amplifying one frequency band more than it should be for the correct tonal balance.

Somewhat remarkably, it is actually fairly easy to get the balance very close to optimum purely by ear. If you have a pair of good headphones, this provides an excellent reference, and any appreciable response deviation in the loudspeaker system can be corrected quite accurately. It is even possible (although not recommended) to use the crossover level controls as a 'tone control' - this can even help make some recordings listenable. Some speakers using passive crossovers provide level controls for mid and high frequencies, and the same can be done with an active system (but with no power loss).

4.4 - What Does All This *Really* Mean?

The crossover / amp / speaker combination has to have the correct gain structure if a flat response is desired, and any variations can be quite audible. The audibility varies with the type of music, and depends a lot on your hearing. In some cases, a slight unbalance can sound better than a perfectly flat system, and can be used to compensate for some room influences, minor driver anomalies or personal preference. Some passively crossed loudspeakers have a L-Pad level control for the tweeter, although these are a lot less common that they once were. If available, this does the same thing as changing the amp gain in an active system.

With a phase-coherent crossover, I have found that the ability to use the crossover gain controls as a 'tone control' seems to work fine, and there are no real anomalies that I have heard (apart from the obvious prominence of the louder frequency band). This is something I often do with my workshop system. This unit runs from my tri-amped, phase-coherent 3-way variable impedance test amplifier (that's a mouthful). I am forever fiddling with the gain structure, amp output impedance and crossover frequencies, and although not ideal (although it has come close with some of the loudspeaker drivers I have there), it nearly always manages to sound much better than it has any right to.

With a conventional passive crossover network, the correct amplitude matching of the loudspeaker drivers is very important, but is usually fixed and cannot be altered. Even with an active system, correct level matching is not just to ensure a flat response, but to ensure that there are no additional phase problems created by the variations. There will nearly always be phase problems with passive crossovers even where the design is very complex, and these errors must create problems in either frequency response or overall accuracy (these don't necessarily coincide, although in an ideal situation they would).

Amplifier loading is another issue that cannot be ignored. The load presented should be essentially resistive (again in this mythical ideal situation), but in reality this is rare. Using electronic crossovers and separate power amplifiers alleviates the issue somewhat, especially for the mid+high section, but the variable reactive load posed by a typical bass reflex type enclosure is always going to cause amplifiers to get a little hotter than expected. Reducing some of the problems by eliminating a passive crossover network (or part thereof) can go a long way towards improving the load seen by the amp, and potentially reducing amplifier intermodulation distortion (in particular). The reduction is largely brought about by making the amp work over a more limited frequency range than normal.

In short, there is no reason for a DIY enthusiast to avoid biamping, since the costs involved are very much lower if you make the crossover and power amplifiers. In addition, it is possible to obtain results that are startlingly good, without the considerable difficulty and expense of tweaking a passive crossover network. The latter can easily cost a hundred dollars or more, and if it's not quite right one can easily spend another hundred dollars trying to get it to sound like it should. By comparison, an active crossover costs between nothing and a few cents to change. Need I say any more? 

5.0 - Passive Biamping (aka Active Biwiring)

Although this is a topic I've mentioned briefly, it needs to be discussed properly. Using two amplifiers and two sets of speaker cables to drive the existing passive crossover is something I call 'passive biamping' or 'active biwiring'. Various websites may claim that it's true biamping, but it's not, never was and never will be.

In some cases users may hear an improvement, but make sure that it really is an improvement and not just a difference. Because you have separate amps driving the two sections of the crossover, you can easily have a level mismatch that leads you to think that the sound is 'better'. The gains of the two amps used must be identical, or the original balance between mid and high will be changed. Naturally the specific frequency depends on where the passive crossover splits the signal. Apart from (usually) a slightly easier load on the amps, both amplifiers still reproduce the full audio bandwidth, so there is no effective power gain.

In general, it is likely that the improvement - assuming there *is* an actual improvement of course - will be small. It will commonly be so small that the additional cost cannot be justified, but this is cold comfort if you've already bought the amplifiers and speaker cables. Speaking of which, make sure that you read the articles on this site about "high end" (rip-off IMO) speaker cables before parting with large sums of money.

For some additional details on the real differences between active an passive systems, see [Active Vs. Passive Crossovers](http://sound.westhost.com/biamp-vs-passive.htm). In particular, loudspeaker damping is usually seriously affected by any passive filter, and commonly right at those frequencies where good damping may help control cone breakup and other unwanted effects.

Conclusion

This really isn't a topic that can be concluded, because there are so many possibilities and variations that there is enough to fill a complete book. Books have already been written about a great deal less, however this is not going to happen and this article (and the others referred to herein) will have to suffice. However, it would be remiss of me - and has been up until now - not to include a very important diagram.

The diagram below was originally produced by Altec, and shows the musical scale, our range of hearing, and various instruments and their most prominent harmonics. I added guitar to the diagram - when it was originally produced the guitar was never taken seriously. I don't have the original publication date, but I recall having seen the diagram a great many years ago. My guess is that the first publication dates from some time in the 1940s. Concert pitch (A440) is shown highlighted in yellow, Middle C is C4 (261Hz) and a grand piano (88 keys) covers the range from A0 to C8.


Figure 8 - Comparative Ranges Of Human Voice And Musical Instrument Frequencies

The above is based on a scan sent to me from the book "Stereo High Fidelity Speaker Systems", by Art Zuckerman 1978. It has been completely re-drawn and this version is copyright © 2010 Rod Elliott. The original is credited to Altec Sound Products.

It is notable that no harmonics are shown for voice - not because they don't exist, but because voice contains other noises (plosives, sibilance and other non-harmonically related sounds). The sound of a "p" right next to a microphone can generate frequencies below 20Hz, and sibilance can create sound above 20kHz. Many of the other harmonic limits shown are considered to be well shy of reality - cymbals can generate frequencies up to around 100kHz for example, but virtually no-one records them and they are considered redundant by most people. I have not attempted to amend the original in this respect, but preferred to leave it as originally created. Interestingly, it seems that no-one is interested enough to re-create this diagram using modern measurement techniques (pity).

The diagram is interesting and included here because it shows the general balance of musical instruments very well. The 'intelligence band' I referred to in several places on this site covers from roughly 300Hz to 3kHz (D4 to F7). Somewhat surprisingly, with *only* this frequency band, we can discern who is speaking and what is being said, or which instrument is playing - we can even hear the bass! This is as heard through telephones ... fixed line, not mobile (the latter are often digital effects units more than telephones) or small portable AM/FM radios through their little speakers. That our ear-brain combination can gather so much from so little is nothing short of astonishing, so this is an area I like to cover with a single driver if at all possible.

Of course, it is not only an important frequency band, but a difficult one to reproduce accurately. This is the true mid-range band. Mess this up, and your best efforts at speaker building will never sound right.