CHAPTER 3

Crossover Requirements

3.1 General Crossover Requirements

The desirable characteristics of a crossover system are easy to state, but not so easy to achieve in practice. There is a general consensus that there are five principal requirements that apply to all crossovers, be they active or passive, and that they should be ranked in order of importance thus:

1. Adequate flatness of summed amplitude/frequency response on-axis
2. Sufficiently steep rolloff slopes between the output bands
3. Acceptable polar response
4. Acceptable phase response
5. Acceptable group delay behaviour

To some extent, the amount of space devoted in this book to each requirement is dependent on their relative importance as set out in this list.

3.1.1 Adequate Flatness of Summed Amplitude/Frequency Response On-Axis

This requires that the output of each filter is appropriate in both amplitude and phase over a sufficient frequency range so that when summation occurs the overall amplitude/frequency response is flat. Note that this requirement does *not* place restrictions on the phase of the summed result; in many cases this will show considerable phase shift that varies across the audio band. This issue is addressed in requirement 4 below.

Crossovers that sum to a completely flat amplitude response include the first-order crossover, the second-order Linkwitz–Riley crossover, the third-order Butterworth crossover and the fourth-order Linkwitz–Riley crossover. A new addition to this list is the Neville Thiele Method notch crossover.

There are many crossover types that can be made to sum very nearly flat by tweaking the filter cutoff frequencies. For example, the second-order Bessel crossover can be made flat to within ±0.07 dB by using a frequency offset ratio of 1.45 times, and the third-order Linkwitz–Riley crossover can be made flat to within ±0.33 dB by applying a frequency offset ratio of 0.872 times. There is much more on this in Chapter 4.

3.1.2 Sufficiently Steep Rolloff Slopes between the Filter Outputs

The filter rolloff slopes must be fast enough to prevent driver damage. Even a small amount of LF energy can rapidly wreck a tweeter. The slopes must be steep enough to not excite areas of poor drive unit frequency response, such as resonances outside normal band of usage; this applies only to mid-range drive units and tweeters, as the LF drive unit resonance will always be used. In addition, the linearity of drive units is very often worse outside their intended frequency range, so steeper slopes will give less non-linear distortion, and that has got to be a good thing.

Restricting the frequency range sent to each drive unit will also minimise frequency modulation distortion caused by the Doppler effect (see Chapter 2). It is also desirable to make the frequency range over which crossover occurs as narrow as possible to minimise the band over which lobing occurs due to two drive units radiating simultaneously.

Unless specially designed drive units are used, the minimum practical slope is usually considered to be 12 dB/octave which requires a second-order crossover. Steeper slopes such as 18 dB/octave (third-order) and 24 dB/octave (fourth-order) are generally considered to be very desirable; 48 dB/octave (eighth-order) slopes are sometimes used in sound reinforcement.

3.1.3 Acceptable Polar Response

An even and well-spread polar response is desirable because it increases the amount of space in which a good sound is obtained. It is also desirable to avoid a large amount of radiated energy from being directed at the floor in front of the loudspeaker, from which it will reflect and cause unwanted comb-filtering effects by interference. Loudspeakers normally give a good polar response in the horizontal on-axis plane, assuming the drive units are mounted in a vertical line as usual. This, however, causes problems in the vertical plane, for in the crossover region two drive units separated in position are radiating simultaneously, and their outputs will interfere, giving reinforcements and cancellations in the radiation pattern at different angles, known as lobing. This is a result of having two drive units separated in space and there is nothing the crossover designer can do about this except make the crossover frequency range as small as possible.

However, it gets worse. If the crossover outputs to each drive unit are in phase, then the main lobe points forward on the horizontal axis, and stays there. If, however, there is a constant phase shift between the outputs, as for first-order crossovers (90° phase shift) or third-order crossovers (270° phase shift) then the main lobe is tilted toward the drive unit that is phase lagging. This is usually the LF unit, so the main energy is being unhelpfully directed towards the floor. This is a frequency-dependent effect because it only occurs in the crossover region and is at its greatest at the crossover frequency itself. It is sometimes simply called “lobing error.”

It is therefore clear that if we are going to keep the main lobe of the summed acoustic output on the axis it is highly desirable that the lowpass and highpass outputs are in phase in the crossover region [1]. This property is possessed only by second-order crossovers (assuming one output is inverted to get a flat response, otherwise the phase-shift is 180°) and fourth-order crossovers with no inversion. This is one reason for the popularity of the fourth-order Linkwitz–Riley crossover.

All the above depends on the drive unit time-delay compensation being correct; the drive units must be either physically mounted or electrically compensated so that the direct sound from each one arrives at the listener's ear at the same time over the whole of the crossover frequency range. Otherwise, the main lobe will have a frequency-dependent tilt toward the driver with the longest air path to the ear.

A good polar response therefore requires that the crossover outputs be in phase and that the time-delay compensation be correct.

3.1.4 Acceptable Phase Response

An acceptable phase response for the *combined* output is also required. Most crossovers are not linear-phase or minimum phase but have the phase response of a first-order allpass filter, with the phase changing by 180° over the audio band. The best-known of these are the first-order (inverted), the second-order Linkwitz–Riley, and third-order Butterworth crossovers. The fourth-order Linkwitz–Riley crossover has the phase response of a second-order allpass filter, with the phase changing by 360° over the audio band. These phase responses are generally agreed to be inaudible with music signals so the fourth requirement is not too onerous. There is a separate section on the issue of phase perception later in this chapter.

3.1.5 Acceptable Group Delay Behaviour

Group delay is simply a measure of how much a signal is delayed. This is directly connected with an acceptable phase response for the combined output, and in fact the group delay is completely determined by the phase shift. Group delay is mathematically the rate of change of the total phase shift with respect to angular frequency (i.e., frequency measured in radians per second rather than Hertz).

Group delay would be of little interest if it was constant, but as the rate of change of phase varies across the audio band, with the phase response of an allpass filter, the group delay also varies. The change is sometimes smooth, but may show a pronounced peak near the crossover frequency. This variation would, if it was sufficiently severe, cause a time-smearing of acoustical events and would sound truly dreadful, but you must not mistake this with the use of the word "smearing" in hi-fi reviews, where it is purely imaginary.

Table 3.1: Variation of Group Delay Threshold with Frequency

Frequency	Group Delay
500 Hz	3.2 ms
1 kHz	2.0 ms
2 kHz	1.0 ms
4 kHz	1.5 ms
8 kHz	2.0 ms

The thresholds for the perception of group delay variation are well known because of their historical importance on long telephone lines. The most accepted thresholds were given by Blauer and Laws in 1978 [2], and are shown in Table 3.1.

These times are given in milliseconds, and a typical group delay for a 1 kHz crossover would be something like 10 times less. The section on phase perception later in this chapter is concerned with the audibility of allpass filters, and the conclusion is firm that neither their phase-shift nor their group delay can be heard on normal musical signals.

The word "group" is derived from "group velocity" in wave-propagation theory, but for our purposes it is simply the amount by which a signal at a given frequency is delayed.

3.2 Further Requirements for Active Crossovers

In addition to the general requirements for all crossovers given above, there are further special requirements for active crossovers. As explained in Chapter 1, if you are adding an extra unit into the signal path it must be as transparent as possible if overall quality is not to take a step backwards. The ultimate goal is total transparency so that the introduction of the crossover causes no degradation at all.

Some further requirements for active crossovers, in no particular order, are:

1. Negligible extra noise
2. Negligible impairment of system headroom
3. Negligible extra distortion
4. Negligible impairment of frequency response
5. Negligible impairment of reliability

It would be easy to add further requirements to this list, such as no degradation in EMC immunity, or of safety, or a modest power consumption, but these apply to any piece of electrical equipment. Those listed above are pretty clearly the most important.

3.2.1 Negligible Extra Noise

While passive crossovers have many limitations, as described in Chapter 1, they do not add noise to the signal. I am sure someone will now point out that crossover inductors do have some resistance and thus must generate Johnson noise, but I am quite certain that −152 dBu of noise (which is what you would get from a 1 Ω resistance) added to a loudspeaker-level signal is one of the lesser problems facing the audio business.

In contrast, the average active crossover performs its processing at line level, or possibly below, and the relative complex structure of a high-quality crossover may allow lots of opportunities for the signal-to-noise ratio to be degraded. Since active crossovers are placed after the main system volume control, turning down the volume will not turn down its noise contribution and with poor design the result could easily be unwelcome levels of hiss from the loudspeakers.

For this reason I have deemed it important to consider noise performance at every step while describing the varied internal circuit blocks of an active crossover. Chapter 14 describes how to minimise noise by using low-impedance design, by using active gain controls, and by adopting the best order for the stages in the crossover. It also deals with the very important possibility of running the crossover at higher nominal internal levels than the input and output signals, provided the placement of gain controls in the whole sound system makes this feasible, so the circuit noise is relatively lower and we get better signal/noise ratios. The intriguing possibility of running the HF crossover path at a still higher level than the others because of the relatively small amount of energy at the top end of the audio spectrum is also looked at in detail. Still another technique is optimising the order of stages in each crossover path; making sure the lowpass filters come last, after the highpass and allpass delay-compensation filters in the path will mean that the noise from upstream is lowpass filtered and may be reduced by several dB. Finally, Chapter 16 on line inputs demonstrates that the conventional balanced line input stage is a rather noisy beast, and shows a number of ways in which to improve it.

I have attempted to bring all these low-noise techniques together in a demonstration crossover design in Chapter 19 at the end of this book. The measured results for the all−5532 version show a signal/noise ratio of 117.5 dB for the HF path output, 122.2 dB for the MID output, and 127.4 dB for the LF output, which I think proves that using all the above noise reduction techniques together can give some pretty stunning results.

Note however, that this crossover will still be a few dB noisier than a naked power amplifier where the input goes straight to the input transistor pair; such power amplifiers have an Equivalent Input Noise (EIN) of about −120 dBu if well designed [3]. However, almost any sort of balanced input in the power amplifier will reverse this situation and the crossover will produce less noise than the amplifier. This point is examined in more depth at the end of Chapter 19.

3.2.2 Negligible Impairment of System Headroom

As described in the previous section, it can be very advantageous to the noise performance to run an active crossover with elevated internal levels, so long as the placement of the gain controls in the whole system permits this. The scenario you are trying to avoid is having a level control between the crossover and the power amplifier that somehow (not me, guv!) gets turned down so it attenuates excessively. Then somebody turns up the volume control on the preamp or pushes up the mixing desk output faders to compensate. This means there is an excessive level inside the crossover, an unexpected signal peak comes along, and… crunch.

This is not a hard situation to avoid. If the active crossover has output level controls that are essentially gain trims with a limited range then it should not be possible to introduce excessive attenuation. In other cases headroom problems are avoided simply by having one competent person with control over the whole system. In hi-fi applications the maximum input levels are fairly well defined by the FSD of the digital source. In other cases, the mechanical limits of the wax cylinder or the vinyl disc impose a less well-defined but still very real limit. In sound reinforcement applications the input levels are much less predictable, but a combination of control from the mixing desk and the use of compressor/limiters should prevent excessive levels from getting as far as the active crossover.

3.2.3 Negligible Extra Distortion

The most significant source of distortion in the average sound system is either the loudspeakers, or if the signal source is vinyl disc, the process of cutting grooves and then wiggling needles about in them. This has never been and never will be accepted as justification for giving up on the design of very linear preamplifiers and power amplifiers. While progress has been made toward making power amplifiers as distortion-free as small-signal circuitry, there are still major technical challenges to be overcome and at present the most significant source of distortion in the electronic domain is almost always the power amplifier. For this reason it may not be too hard to design a signal path that is significantly more free of distortion, especially the crossover distortion that we all abhor, than the average power amplifier. Nevertheless, as I have said before, we are inserting an extra signal-processing box into the signal path, and it behoves us to make the degradation it introduces as small as economically possible. This is not too hard from the cost point of view as the active crossover, even if built to the highest standards, is likely to be much cheaper than the extra power amplifiers required for multi-way operation.

3.2.4 Negligible Impairment of Frequency Response

This may seem like a strange requirement in a piece of equipment whose whole *raison d'être* is radical modification of the spectrum of the signals it handles, but here I want to distinguish between the frequency response modifications you want and those you don't. Even if we assume that an active crossover has a well-conceived filter structure, correct filter characteristics, suitably low component sensitivities, and is built with components of accurate value, it would still be possible to degrade the frequency response by using an over-aggressive ultrasonic filter or an inappropriate subsonic filter, and these stages should get their full share of attention.

3.2.5 Negligible Impairment of Reliability

There is nothing that upsets the paying customer more than equipment that stops working (OK, if it sets fire to the house that would probably annoy them rather more), so I make no apology for putting this rather general requirement in. We are dealing with opamp circuitry using modest supply rail voltages, the general levels of voltage and current are low, and the opamps even have internal overload protection. So as long as the designer knows what they're about there is no reason why any components should be much stressed. There are a couple of not-quite-obvious things that could go wrong in the power supply, such as regulator heatsinks that are normally OK but prove to be too small when the mains is 10% high in a hot country, or ill-conceived decoupling capacitors that cause the supply to fail to start on an unpredictable basis. These design landmines are dealt with in Chapter 18 on power supply design.

3.3 Linear Phase

A linear-phase crossover has a combined output phase-shifted by an amount proportional to frequency; in other words it introduces a pure time-delay only, and the group delay is constant. Non-linear-phase crossovers have phase-shifts that change non-linearly with frequency and act like allpass filters; for this reason they are often called allpass crossovers. Now while linear phase is clearly desirable on a purely theoretical basis, in that it makes the crossover more transparent and closer to perfection, it is difficult to achieve. There is certainly no settled consensus that it is necessary for a good acoustic performance, and the bulk of evidence is that it is simply not necessary for satisfactory results on normal music. There is more on this issue in the section below on phase perception.

The best-known crossover types with linear phase are first-order non-inverted crossovers, filler-driver crossovers, and subtractive crossovers with time delay such as those put forward by Lipshitz and Vanderkooy in 1983 [1].

3.4 Minimum Phase

Minimum phase, or minimal phase, is a term that is sometimes confused with linear phase. In fact they are not only not the same thing, but almost opposites. A minimum-phase system, in our case a filter, has the minimum phase shift possible to get the amplitude/frequency response it shows. A minimum-phase filter is also one where the phase/frequency response can be mathematically derived from the amplitude/frequency response, and vice-versa. Furthermore, the effect of a minimum-phase filter can be completely cancelled out in both phase and amplitude by using a reciprocal filter that has the opposite effect. A good example of this is given in Figure 11.1 in Chapter 11 on equalisation, where it is demonstrated that a peaking equaliser can be exactly cancelled out by a dip equaliser, and a square-wave put through them both is reconstructed.

There are, as you might expect, several much more precise mathematical ways of defining the minimum-phase condition [4], but they are not helpful here. In general you cannot say that a minimum-phase filter is better than a non-minimum-phase filter, as it depends what you are trying to do with said filter.

Most crossover filters, such as highpass and lowpass types in their various kinds are inherently minimum phase. The classic exception to this is the allpass filter used for time-delay correction. Since an accurate allpass filter has a completely flat amplitude/frequency response, you cannot deduce anything at all from it about the phase/frequency response. The phase-shift of an allpass filter in fact varies strongly with frequency, as described in Chapter 10 on time-delay compensation, but the amplitude/frequency response gives you no clue to that. It is also impossible to undo the effect of an allpass filter because its particular phase-shift characteristics give a constant delay at suitably low frequencies, and you cannot make a filter that has a negative delay. That would mean foretelling the future, and on the whole would probably not be a good thing.

The first-order crossover is minimum phase when its outputs are summed normally; it has a flat phase plot at 0°. If one output is inverted, however, while the SPL and power responses are still flat, the summed output has a first-order allpass phase response, the phase swinging from 0° to −180° over the frequency range. It is therefore no longer minimum-phase.

As we just noted a linear-phase crossover acts as a pure time delay, and so cannot be minimum-phase. You cannot, however, say that a crossover which includes allpass filters for time-delay correction can never be minimum phase, because they are correcting for physical misalignments and what counts is the summed signal at the ear of the listener.

3.5 Absolute Phase

Another phase issue is the perception of absolute phase. In other words, if the polarity of a signal is inverted (it is not relevant whether it is heard via a single or multi-way loudspeaker) does it sound different? The answer is yes, providing you use a single tone with a markedly asymmetrical waveforms, such as a half-wave rectified sinewave or a single unaccompanied human voice. Otherwise, with more complex signals such as music, no difference is heard.

Obviously an active crossover must have all its outputs in the correct phase with each other (in some cases correct means phase-inverted) or dire response errors will result, but it is also necessary to make sure that the outputs are in the correct phase relationship to the crossover input signal.

Almost all hi-fi equipment such as preamplifiers and power amplifiers are now designed to preserve absolute phase. Mixing consoles have always been so designed to prevent unwanted cancellation effects on mixing signals.

3.6 Phase Perception

Some of the crossovers described in this book have quite dramatic phase changes in the summed output around the crossover points, so the sensitivity of human hearing to phase shift is an important consideration. If the phase-shift is proportional to frequency then the group delay is constant with frequency and this is a linear-phase system, as described above; we just get a pure time-delay with no audible consequences. However, in most cases the phase-shift is not remotely proportional to frequency, and so the group delay varies with frequency. This is sometimes called group delay distortion, which is perhaps not ideal as 'distortion' implies non-linearity to most people, while here we are talking about a linear process.

Most of the components in the microphone-recording-loudspeaker chain are minimum-phase; they impose only the phase-shift that would be expected and can be predicted from their amplitude/frequency response. The great exception to this is… the multi-way loudspeaker. The other great exception was the analogue magnetic tape-recorder, which showed rapid phase-changes at the bottom of the audio spectrum, usually going several times round the clock [5]. Fortunately we don't need to worry about *that* any more.

We are, however, going to have multi-way loudspeaker systems around for the foreseeable future, and most of them have allpass crossovers. Clearly an understanding of what degradation, if any, this allpass behaviour causes is vital. Much experimentation has been done and there is only space for a summary here.

One of the earliest findings on phase perception was Ohm's Law. No, not that one, but Ohm's *Other* Law, which is usually called Ohm's Acoustic Law, and was proposed in 1843 [6]. In its original form it simply said that a musical sound is perceived by the ear as a set of sinusoidal harmonics. The great researcher Hermann von Helmholtz extended it in the 1860s into what today is known as Ohm's Acoustic Law, by stating that the timbre of musical tone depends solely on the number and relative level of its harmonics, and *not* on their relative phases. This is a good start, but does not ensure the inaudibility of an allpass response.

An important paper on the audibility of midrange phase distortion was published by Lipshitz, Pocock and Vanderkooy in 1982 [7] and they summarised their conclusions as follows:

1. Quite small phase non-linearities can be audible using suitable test signals.
2. Phase audibility is far more pronounced when using headphones instead of loudspeakers.
3. Simple acoustic signals generated in an anechoic environment show clear phase audibility when headphones are used.
4. On normal music or speech signals phase distortion is not generally audible.

At the end of the abstract of their paper the authors say: "It is stressed that none of these experiments thus far has indicated a present requirement for phase linearity in loudspeakers for the reproduction of music and speech." James Moir also reached the same conclusion [8].

An interesting paper on the audibility of second-order allpass filters was published in 2007 [9], which describes a perception of "ringing" due to the exponentially decaying sinewave in the impulse response of high Q all-pass filters (For example Q = 10). It was found that isolated clicks show this effect best, while it was much more difficult to detect, if audible at all, with test signals such as speech, music, or random noise. That is the usual finding in this sort of experiment—that only isolated clicks show any audible difference. While we learn that high-Q allpass filters should be avoided in crossover design, I think most people would have thought that was the case anyway.

Siegfried Linkwitz has done listening tests where either a first-order allpass filter, a second-order allpass filter (both at 100 Hz), or a direct connection could be switched into the audio path [10]. These filters have similar phase characteristics to allpass crossovers and cause gross visible distortions of a square waveform, but are in practice inaudible. He reports "I have not found a signal for which I can hear a difference. This seems to confirm Ohm's Acoustic Law that we do not hear waveform distortion."

If we now consider the findings of neurophysiologists, we note that the auditory nerves do not fire in synchrony with the sound waveform above 2 kHz; so unless some truly subtle encoding is going on (and there is no reason to suppose that there is), then perception of phase above this frequency would appear to be inherently impossible.

Having said this, it should not be supposed that the ear operates simply as a spectrum analyser. This is known not to be the case. A classic demonstration of this is the phenomenon of "beats." If a 1000-Hz tone and a 1005-Hz tone are applied to the ear together, it is common knowledge that a pulsation at 5 Hz is heard. There is no actual physical component at 5 Hz, as summing the two tones is a linear process. (If instead the two tones were multiplied, as in a radio mixer stage, there *would* be new components generated) Likewise non-linearity in the ear itself can be ruled out if appropriate levels are used.

What the brain is actually responding to is the envelope or peak amplitude of the combined tones, which does indeed go up and down at 5 Hz as the phase relationship between the two waveforms continuously changes. Thus the ear is in this case acting more like an oscilloscope than a spectrum analyser.

It does not however seem to work as a phase-sensitive detector.

The conclusion we can draw is that a crossover whose summed phase response is that of a first-order or second-order allpass filter is wholly acceptable. This obviously implies that a group delay characteristic that emulates a first- or second-order allpass filter is also completely acceptable.

3.7 Target Functions

A target function for a loudspeaker system is the combined crossover and loudspeaker response that you are aiming for. Drive units are hopefully fairly flat over the frequency range that we hope to use them, but if this is not the case, then their response obviously has to be taken into account. Response irregularities may be corrected by equalisation (see Chapter 11), performed either by adding dedicated equalisation stages to the relevant crossover path or by modifying the characteristics of the crossover filters. The latter uses less hardware but is much more difficult to understand unless properly documented.

In some cases the inherent properties of the drive unit and the enclosure may form part of the target function. For example, a suitably damped LF unit and enclosure will have a second-order Butterworth-type maximally flat rolloff at 12 dB/octave. If this is combined with a second-order Butterworth highpass filter in the crossover, then this makes up a Linkwitz–Riley fourth-order alignment which can be used to crossover to a separate subwoofer.

References

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