ASHLY

ELECTRONIC CROSSOVERS OPERATING INSTRUCTIONS

ASHLY AUDIO, INC.

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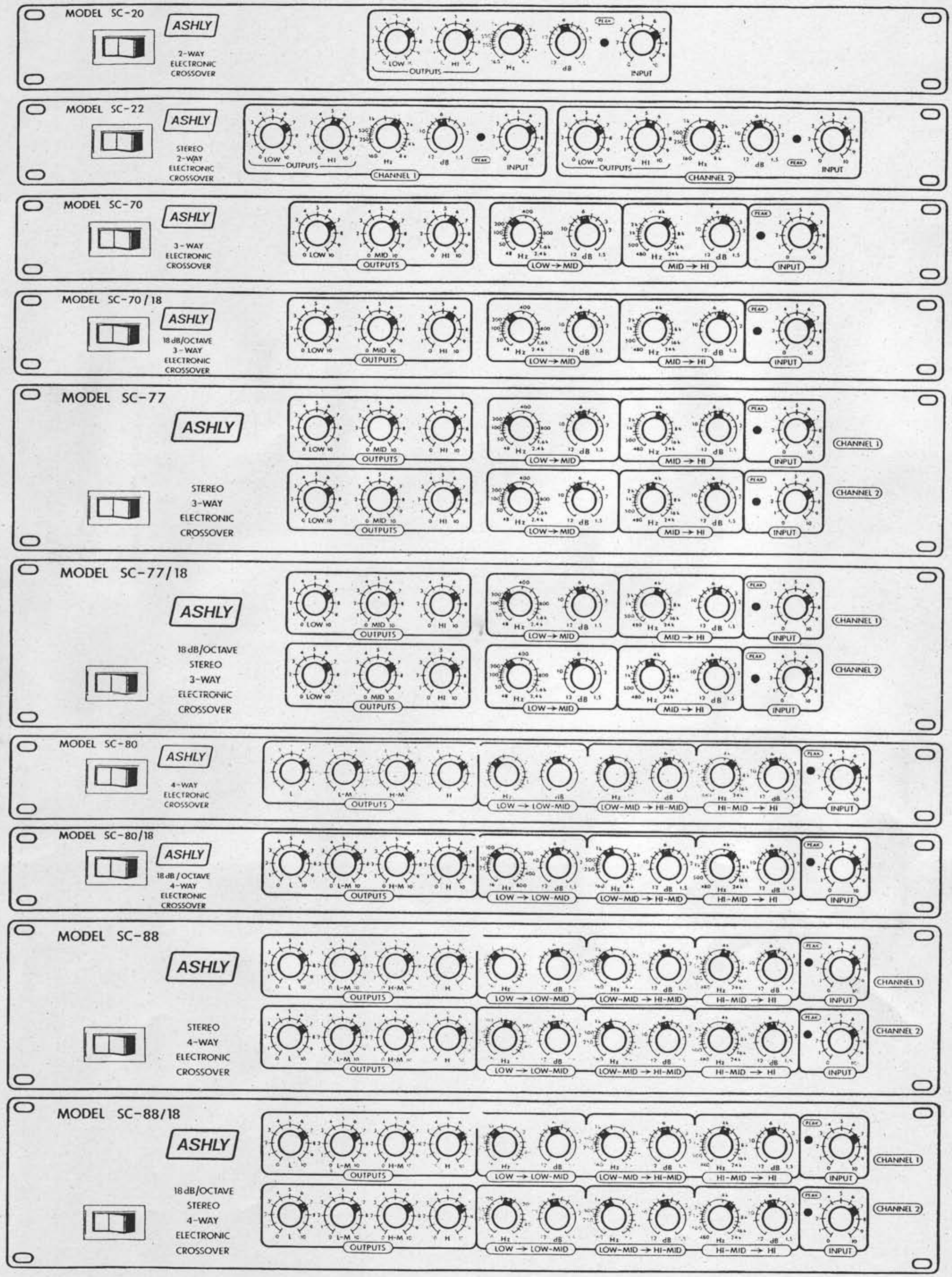


Figure 1 Front Panel Illustrations

INTRODUCTION

Your Ashly crossover is the product of an intensive research effort which combined a re-examination of traditional crossover theory with practical field testing to prove the new design. Over the past several years, a number of refinements and new models have been added to our crossover series, but the original design goals have remained the same: to produce a crossover which is sonically accurate, is flexible enough to adapt to a wide variety of systems, and which affords maximum protection for speakers and drivers.

Ashly Audio now manufactures the world's most extensive line of electronic crossovers for the professional audio industry, from a mono 2-way to our stereo 4-way, with a choice of either 12dB per octave or 18dB per octave slopes. All Ashly crossovers, from the simplest to the largest, are based upon the same powerful state-variable filter circuit, thus insuring superior performance from each model. Our crossovers offer a number of useful and unusual features, including thump-free turn-on without the use of relays, a damping (rolloff) control that functions much like an equalizer centered right at the crossover frequency, and a special output stage which maintains low noise at any level setting. Like other Ashly products your crossover features low noise and distortion, active balanced inputs, a peak level indicator, a precision regulated power supply, protection against abnormal input or output conditions, and rugged mechanical construction. Conservative design and an unusually thorough procedure for quality control have earned Ashly a reputation for dependability in the recording, sound reinforcement, and broadcast fields.

Due to the similarity of the operating controls on our various crossovers, we have prepared this one manual to cover all the models in our crossover line. Specific notes on each model will be found in the Applications section. Many owners are unaware of the ease with which you can reconfigure your crossover for non-standard operation. For example, a 4-way crossover can easily be used as a mono 2-way with sweepable subsonic and ultrasonic cutoff filters, and a stereo 3-way can function as a mono 5-way crossover. No internal modifications are necessary, and audio performance is not compromised. These operating modes are explained in detail beginning on page 8.

In addition, this manual covers normal setup and operation, answers some frequently asked questions about our crossovers, offers trouble-shooting tips, and, in a special section, covers basic crossover theory.

Please take the time to read this manual before operating your crossover.

SECURITY COVERS

For installations where it is desirable to protect the front panel controls from tampering or accidental misadjustment, use the Ashly security cover, which is available in both single and double rack space sizes. These covers feature rugged formed and welded steel construction with a clear plexiglass window that allows you to see all control settings. Installation is simple and does not require removal of the equipment from your rack. See your Ashly dealer for details.

UNPACKING

As a part of our system of quality control every Ashly product is carefully inspected before leaving the factory to ensure flawless appearance. After unpacking, please inspect for any physical damage. Save the shipping carton and all packing materials, as they were carefully designed to reduce to a minimum the possibility of transportation damage should the unit again require packing and shipping. In the event that damage has occurred, immediately notify your dealer so that a written claim to cover the damages can be initiated.

THE RIGHT TO ANY CLAIM AGAINST A PUBLIC CARRIER CAN BE FORFEITED IF THE CARRIER IS NOT NOTIFIED PROMPTLY AND IF THE SHIPPING CARTON AND PACKING MATERIALS ARE NOT AVAILABLE FOR INSPECTION BY THE CARRIER. SAVE ALL PACKING MATERIALS UNTIL THE CLAIM HAS BEEN SETTLED.

INPUT, OUTPUT, AND POWER CONNECTIONS

This crossover should be connected to a 3 wire grounded outlet supplying 120 Volts, 50-60 Hz. Power consumption is 12 watts.

The INPUT is a 10K ohm active balanced type on a standard stereo phone plug. The (+) or in-phase connection is on the tip and the (-) or out-of-phase connection is on the ring. When feeding the crossover from unbalanced sources, connect the signal hot to the tip (+) and the signal ground to the ring (-). To use the input as a common unbalanced type, simply use a mono phone plug in the usual way. (See Definition Of Terms, "Wiring", page 35.)

The OUTPUT is a low impedance type (typically about 50 ohms). This should be terminated in a load of 600 ohms or higher to realize maximum headroom. The output can be used as a balanced type by using a stereo phone plug wired in the same manner as the input or as an unbalanced type by using a mono phone plug.

Stereo 3-way and 4-way crossovers have an additional output jack labeled MONO LOW OUT. This is a summed mono output taken from the low frequency outputs of each channel. It is typically used to drive a sub-woofer system.

EXPLANATION OF OPERATING CONTROLS

Every Ashly crossover has the following controls: An INPUT level control, a CROSSOVER FREQUENCY (Hz) control, a ROLLOFF (dB) control, OUTPUT level controls, and a PEAK indicator LED.

INPUT LEVEL CONTROL

The Input control allows a wide range of nominal system levels to be used. For unity gain operation, set it to 7.

CROSSOVER FREQUENCY CONTROL

This infinitely-variable control allows you to select an appropriate crossover point for your speakers. Turning the knob clockwise moves the crossover point to a higher frequency, while turning it counterclockwise moves it to a lower frequency.

Crossover frequencies are marked on standard ISO 1/3 octave center frequencies with every octave calibrated. Calibration accuracy is very good, typically within 1/3 octave or better. If greater accuracy than this is necessary, measurement of the actual crossover frequency with an accurate oscillator and/or frequency counter is suggested.

The choice of crossover frequencies depends on the type of speakers being used, personal taste, room acoustics, and many other factors. Experiment to see what works best for you.

CAUTION: **High frequency compression drivers may be destroyed by the use of too low a crossover frequency. Follow the driver manufacturer's recommendations carefully.**

ROLLOFF (dB) CONTROL

This control, found adjacent to the crossover frequency control, adjusts the damping of the filter at the crossover point, affecting the response shape of the filters. The dial calibrations (1.5, 3, 6, 10, 12) refer to the amount of attenuation effected by the filters at the crossover frequency, i.e., a setting of 3 means that the filter's response is "rolled off 3dB at the crossover point", which describes Butterworth filter response.

In actual use, the rolloff control acts much like an equalizer tuned exactly to the crossover frequency. For example, with our 12dB per octave models, a setting of 6 on the control will give you flat response through the crossover region. Turning the control counterclockwise will cause a dip in the response at the crossover frequency, while turning it clockwise will cause a gentle peak in the response. On our 18dB per octave models, a setting of 3 will yield flat summed response.

The purpose of the control is to help offset the inaccuracies inherent in typical loudspeaker response, thereby helping you to achieve a flat system response. If you use a spectrum analyzer to set up your system, adjust the rolloff control to obtain the flattest response through the crossover region before any equalization is applied to the system. If adjusting by ear, we recommend an initial setting of 6 for our 12dB per octave models, and a setting of 3 for our 18dB per octave models. Then adjust from this point if the system appears to have an excess or deficiency of response at the crossover point.

NOTE: **The rolloff control is not a "slope" control. A 12dB/octave crossover will always have a slope of 12dB/octave regardless of the setting of this control. Likewise, an 18dB/octave crossover will always have a slope of 18dB/octave. The dB control only affects filter response shape in the immediate vicinity of the crossover frequency; the ultimate slope of the crossover is a fixed parameter.**

OUTPUT LEVEL CONTROLS

In a typical setup, power amplifiers can be run "full-on", with level control being accomplished at the crossover. Adjust these controls for best system balance. Note that horn and compression driver combinations are much more efficient than cone speakers, often by 12-20dB. When cones and compression drivers are used together, you should expect a much lower level setting for the horns (all other factors being equal) to compensate for this difference and obtain proper balance.

PEAK LED

Ashly crossovers feature a peak detection circuit which monitors signal levels at several critical points in the crossover: input, filters, and outputs. The LED will flash when signal levels of +14dBV are reached anywhere in the crossover. Since our crossovers have a nominal 20dB of headroom referenced to a standard operating level of 0dBV (.77 Volts), a flashing LED warns you that you are only 6dB away from clipping.

Choice of a "nominal" system operating level is really up to the operator. You may want to run the system "hot", with the peak LED flashing frequently, or you may want to run the system more conservatively, with the LED flashing seldom or never.

Obviously, the hotter your nominal level, the less system headroom you will have. For initial setup, it is recommended that you adjust your level controls so that the LED flashes only on the loudest signal peaks, if at all.

Since peak levels are monitored at several key circuit points, the peak LED can be used to isolate the source of any overload. If the LED flashes even though all input and output levels are turned down, then the signal being fed to the crossover is excessively hot. In most cases this means you should go back and turn down your mixer. If the LED flashes when you turn the Input level control up (with the outputs still turned down), the overload is occurring in the filter sections, and you should back the Input level down a bit. If the LED first flashes when you turn the output level controls up, then the overload is occurring in the output stage, and you need to turn the output controls down.

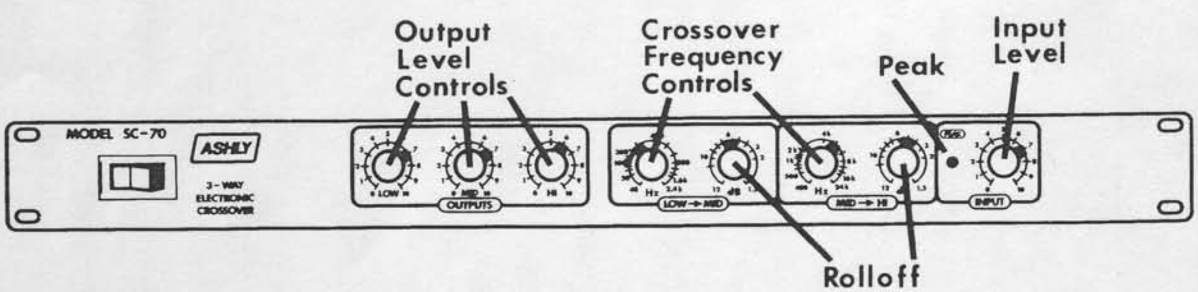


Figure 2 An SC-70 3-way crossover, showing front panel controls generic to all Ashly crossovers.

NORMAL SETUP GUIDELINES

Speaker Placement

In order for you to obtain maximum benefit from your crossover, a correctly assembled speaker array is a necessity. Low frequency speakers should be grouped as closely together as possible and mid and high frequency horns should be stacked on top of one another, not side by side. This arrangement insures a tight vertical pattern combined with wide horizontal dispersion free from high frequency lobing. Speakers in the array should be arranged so the drivers and cones are all in the same plane (equal distance from the listener).

Phasing

It is unnecessary to reverse the polarity of any of the outputs of an Ashly crossover, as all necessary inversions have been done internally. Simply stack your speakers and wire positive to positive, negative to negative on all outputs.

With the multitude of different brand amplifiers and speakers used together, phasing of all components of multi-band systems can be very confusing. Only extreme care in the hookup of these components will insure proper phasing. It is important not only to keep all the speakers within each band in phase with each other but to keep all bands of the system in phase as well. If this is not done, loss of level and pattern control at the crossover frequency will result.

Phase of cone speakers can be checked by connecting a 1.5 Volt battery to the speaker and observing which way the cone moves. The most common convention is that (+) voltage on the (+) terminal moves the cone forward. A notable exception to this convention is JBL. A (+) voltage on the red terminal of a JBL speaker moves the cone backward. If all your speakers are the same brand, just connect them all the same way; if not, it's best to test. Unfortunately, compression drivers cannot be tested this way. We do know that Altec, Emilar, and Electrovoice agree and that JBL is reversed from their standard.

Most power amplifiers have their output in-phase with the input and it is generally not a problem to mix brands.

CONNECTIONS FOR NORMAL OPERATION

Connecting a crossover to your system is straightforward if you are using it in its normal configuration--the output of your mixer connects to the input of the crossover and the outputs are connected to appropriate power amplifiers as shown in figure 3.

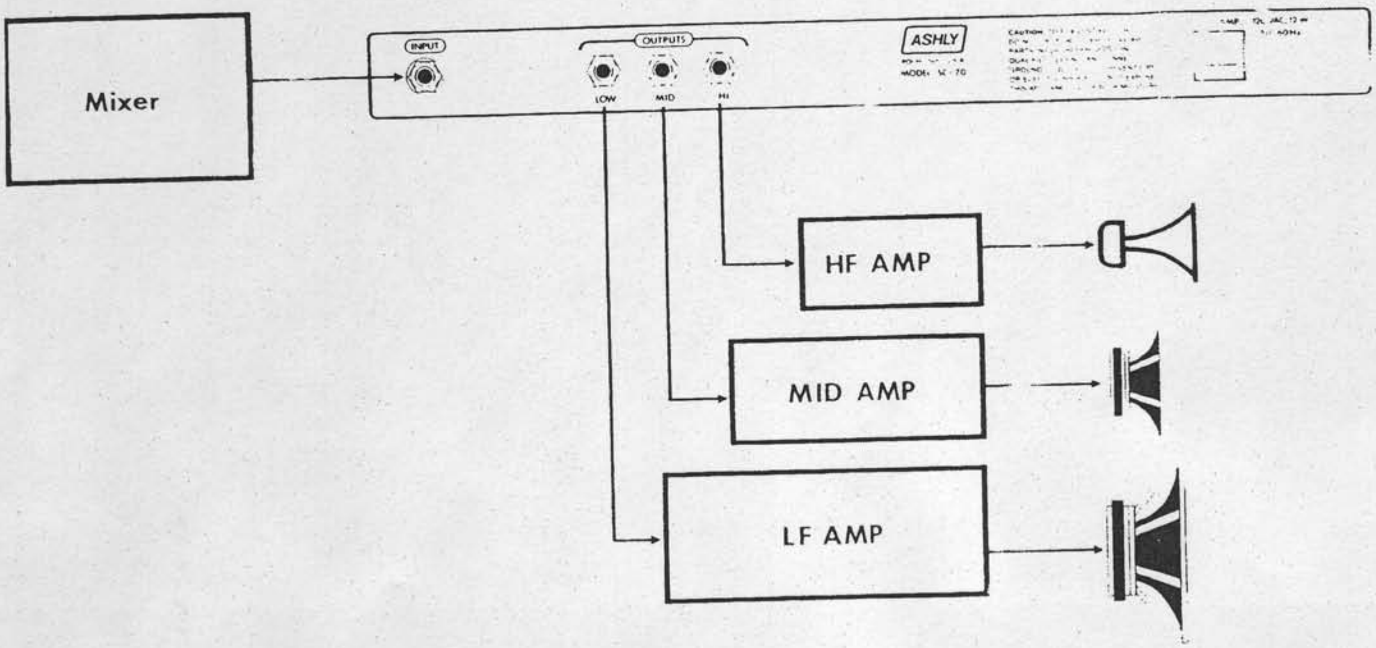


Figure 3 SC-70 3-way crossover showing typical plug-in arrangement for normal operation.

MONO LOW OUTPUT

Ashly stereo 3-way and stereo 4-way crossovers have an additional output jack labeled MONO LOW OUT. This output represents the sum of the low frequency outputs of channels one and two, and is typically used for driving a sub-woofer system. If you are using the mono low output, you may also use the normal low frequency outputs to drive other speaker systems; any or all of the low frequency outputs may be used at any given time without fear of interaction between outputs.

SPECIAL APPLICATIONS

All Ashly crossovers, with the exception of the mono SC-20, may be operated in some configuration other than the normal mode. For example, a stereo 4-way crossover can also function as a mono 7-way, and a mono 3-way can function as a mono 2-way. The following pages show how to connect your crossover for these special situations, with suggested front-panel settings.

NOTES:

Actual crossover frequency selection will depend on specific loudspeakers and personal taste. The settings shown are for illustration only.

For the sake of simplicity, only our 12dB per octave models are shown. Initial setup of both 12 and 18dB/octave models will be the same with the exception of the rolloff (dB) control settings. You'll see that on 12 dB per octave models, we've recommended a setting of "6" on the rolloff control for all crossover points, and a setting of "3" for all low-pass and high-pass cutoff frequencies. For 18dB per octave models, just set all rolloff controls to "3", regardless of function.

SC-20 MONO 2-WAY

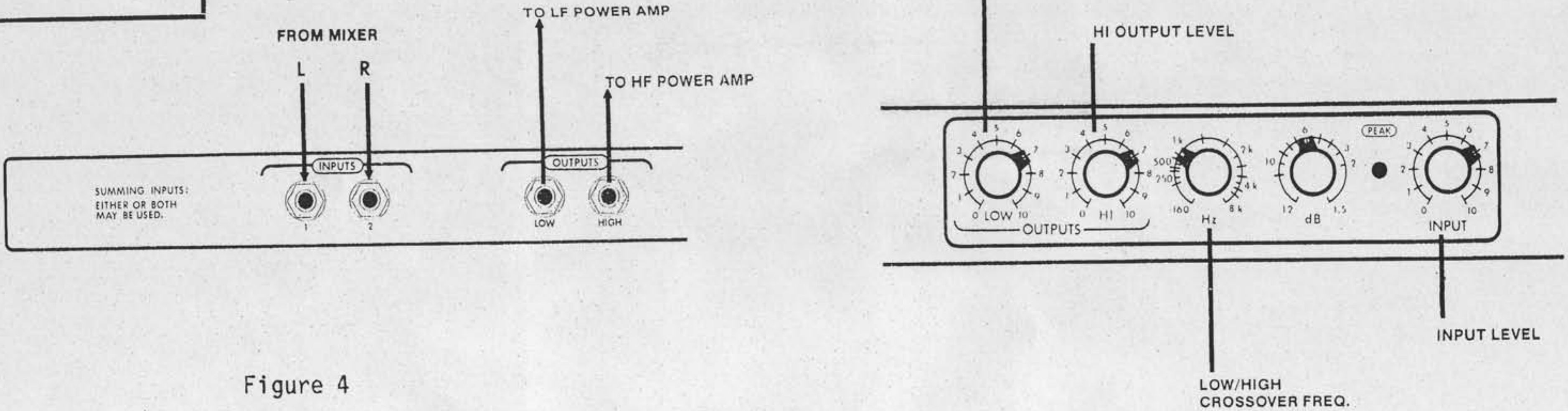


Figure 4

The SC-20 has a feature not found on our other crossovers: summing inputs. When used in conjunction with a stereo mixing console, you can use the left and right channels of the mixer to control two submixes, and then sum these outputs at the crossover. For mono mixers, just plug into either of the inputs, since they are identical.

SC-22 STEREO 2-WAY USED AS A MONO 3-WAY

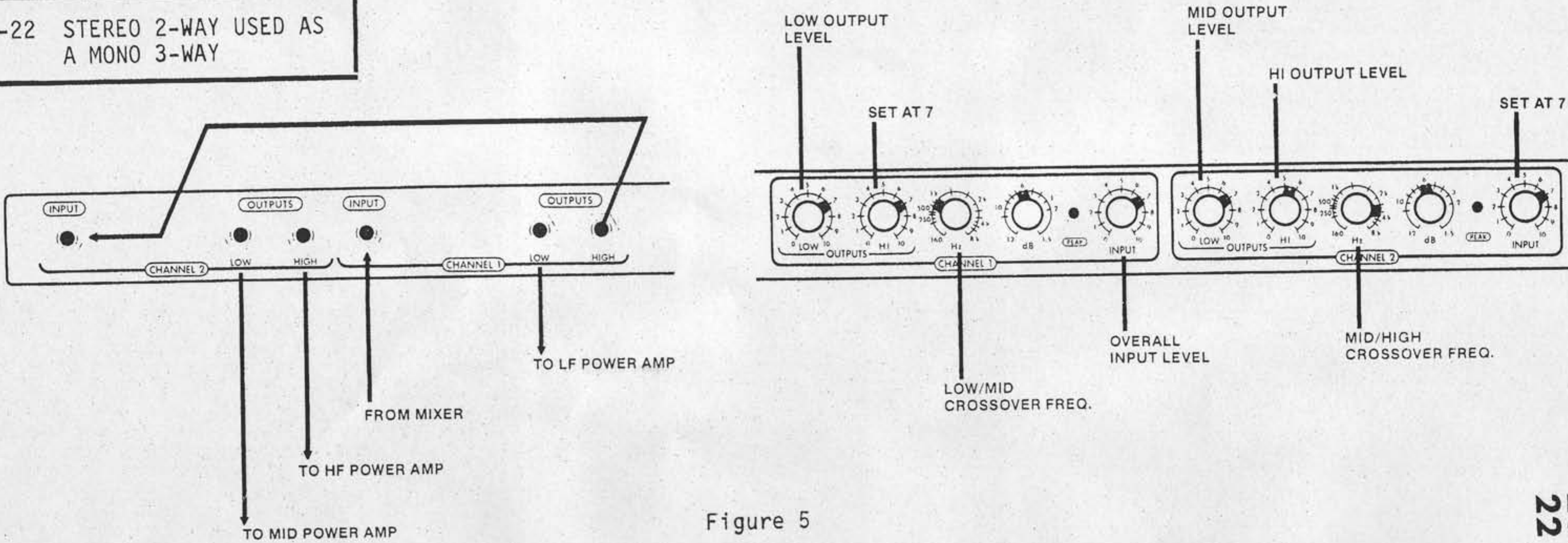


Figure 5

20
22

SC-70 USED AS A MONO 2-WAY
WITH TUNABLE SUBSONIC FILTER

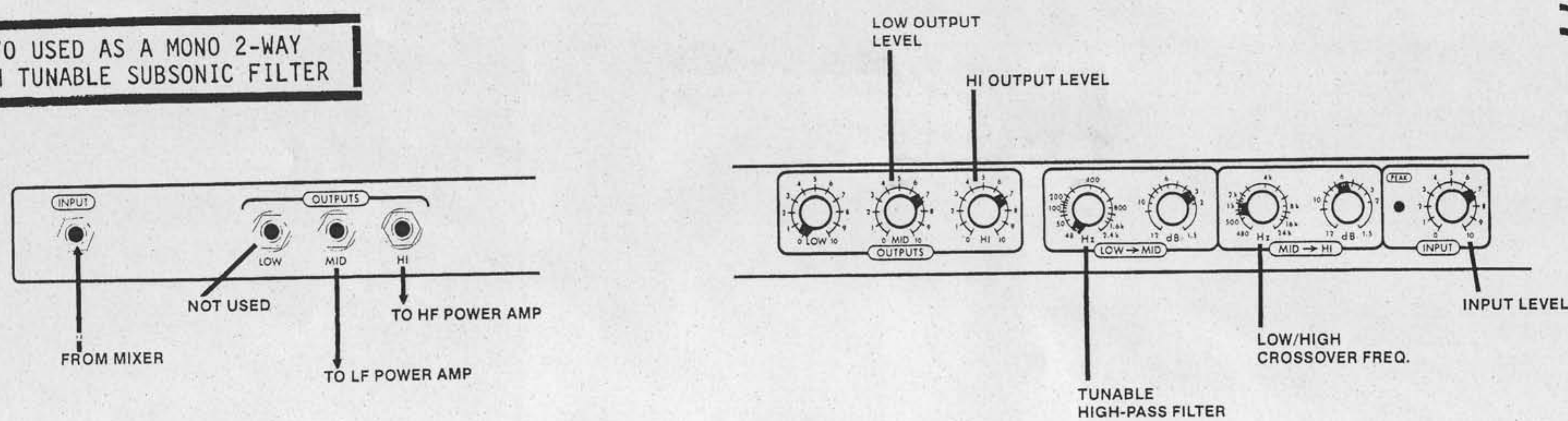


Figure 6

SC-70 USED AS A MONO 2-WAY WITH
TUNABLE ULTRASONIC FILTER

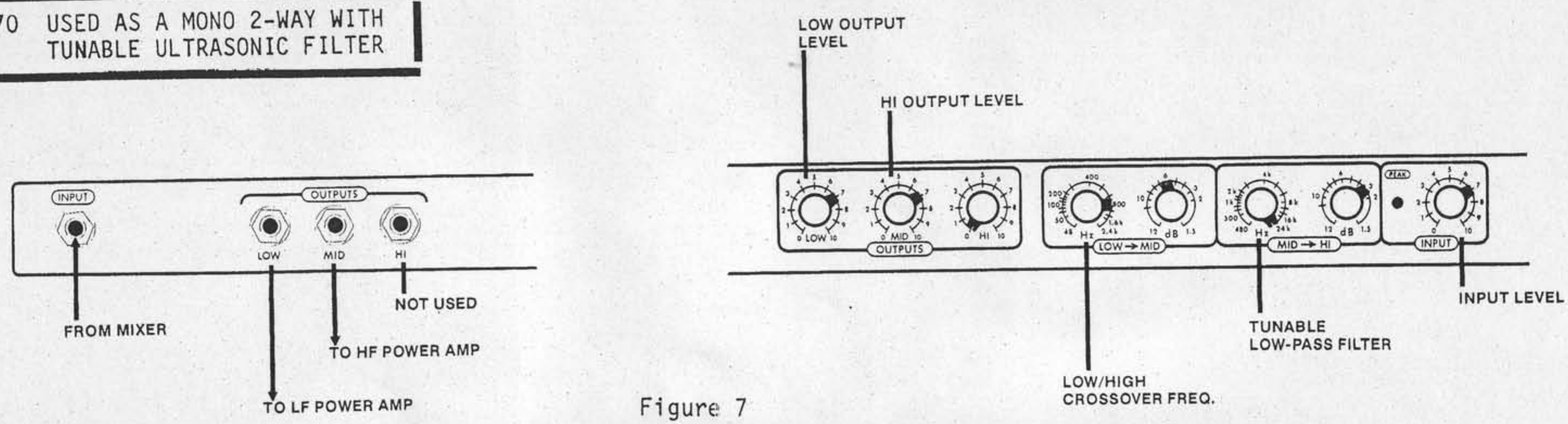


Figure 7

SC-77 USED AS A MONO OR STEREO 2-WAY
WITH TUNABLE SUBSONIC FILTER

See Figure 6

SC-77 USED AS A MONO OR STEREO 2-WAY
WITH TUNABLE ULTRASONIC FILTER

See Figure 7

SC-77 USED AS A MONO 4-WAY WITH TUNABLE SUBSONIC FILTER

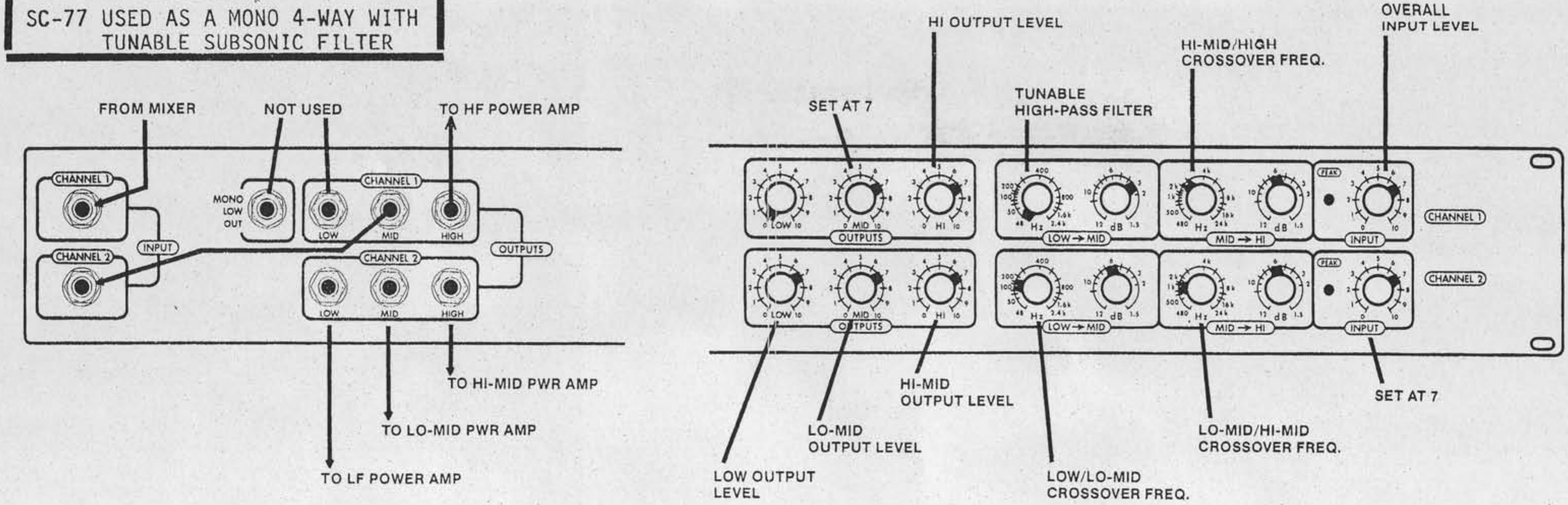


Figure 8

SC-77 USED AS A MONO 4-WAY WITH TUNABLE ULTRASONIC FILTER

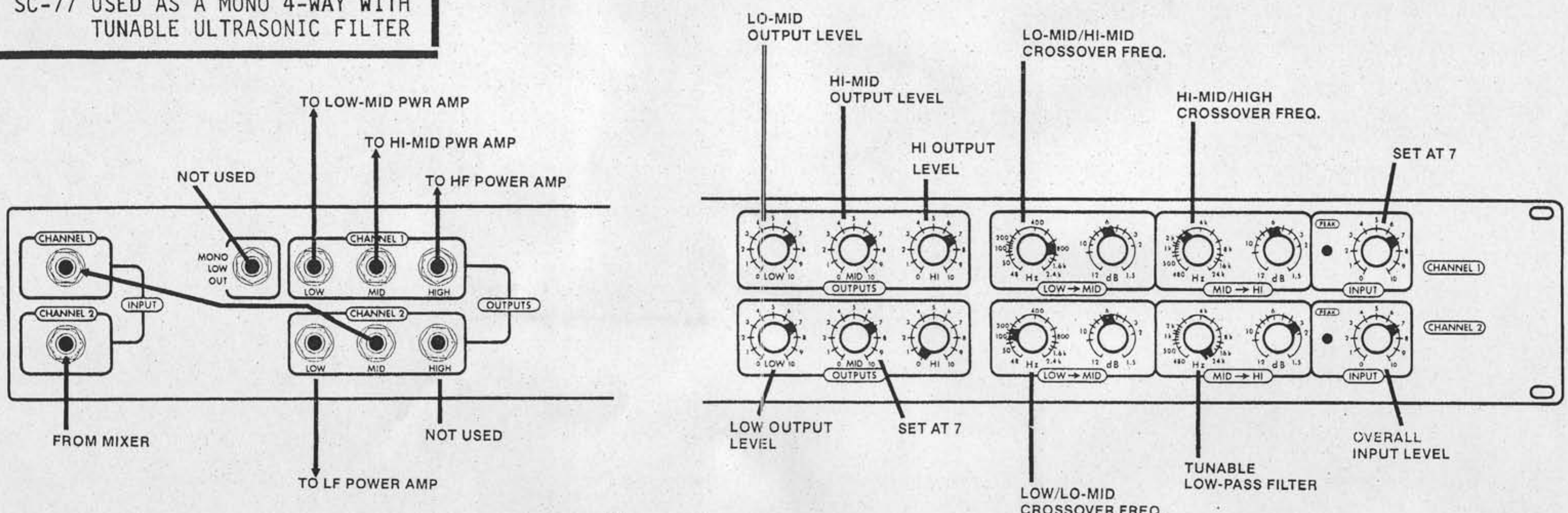


Figure 9

SC-77 USED AS A MONO 5-WAY

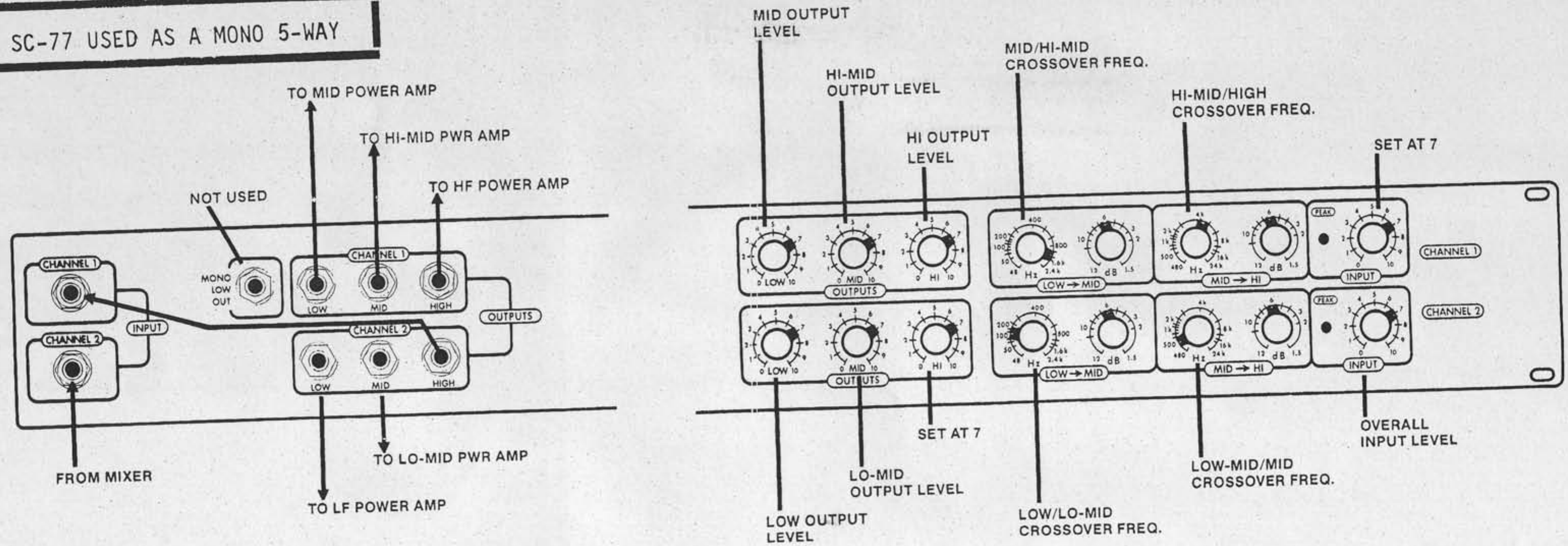


Figure 10

SC-80 USED AS A MONO 2-WAY WITH TUNABLE SUBSONIC AND ULTRASONIC FILTERS

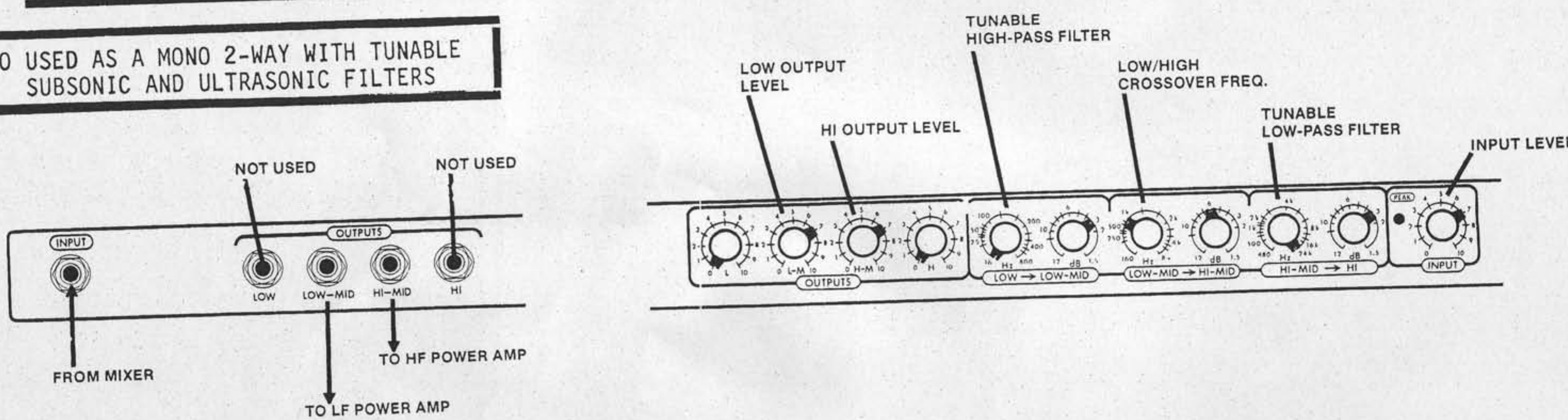


Figure 11

SC-80 USED AS A MONO 3-WAY WITH TUNABLE SUBSONIC FILTER

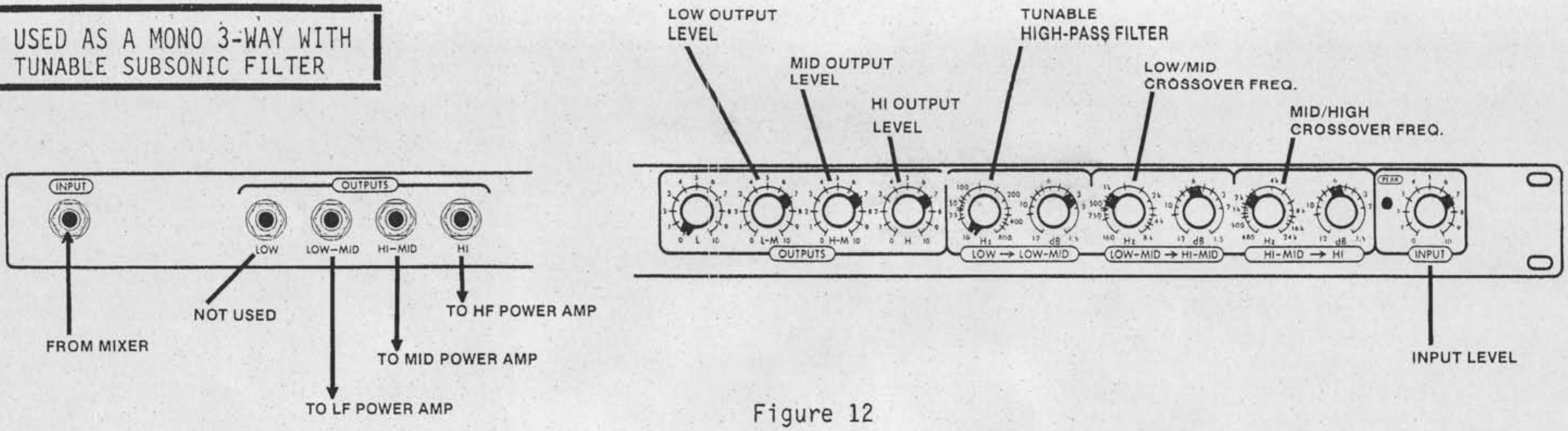


Figure 12

SC-80 USED AS A MONO 3-WAY WITH TUNABLE ULTRASONIC FILTER

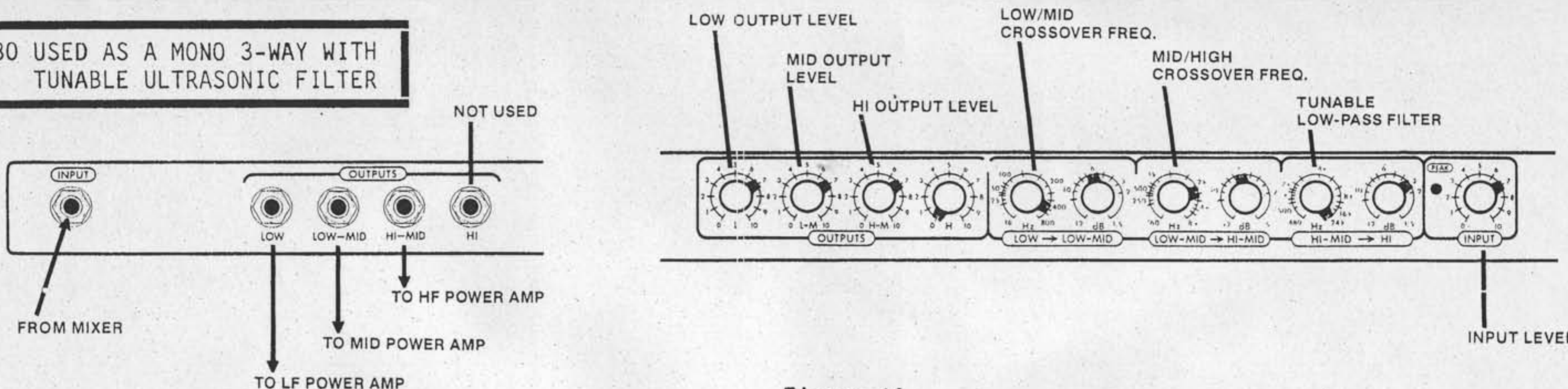


Figure 13

SC-88 USED AS A MONO OR STEREO 2-WAY WITH TUNABLE SUBSONIC AND ULTRASONIC FILTERS

See Figure 11

SC-88 USED AS A MONO OR STEREO 3-WAY WITH TUNABLE SUBSONIC FILTER

See Figure 12

SC-88 USED AS A MONO OR STEREO 3-WAY WITH TUNABLE ULTRASONIC FILTER

SC-88 USED AS A MONO 5-WAY WITH TUNABLE SUBSONIC AND ULTRASONIC FILTER

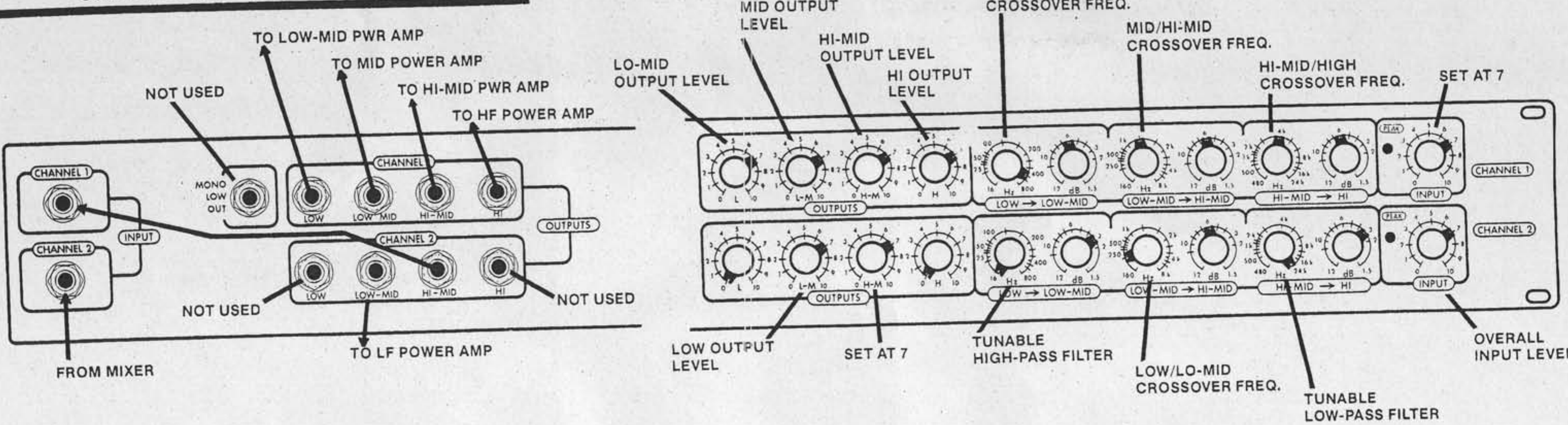


Figure 14 Use the above version when crossover points will be relatively high. The lowest LOW/LO-MID crossover frequency of 160 Hz may restrict the use of this configuration. If lower crossover frequencies are needed, use the configuration shown in figure 15 below.

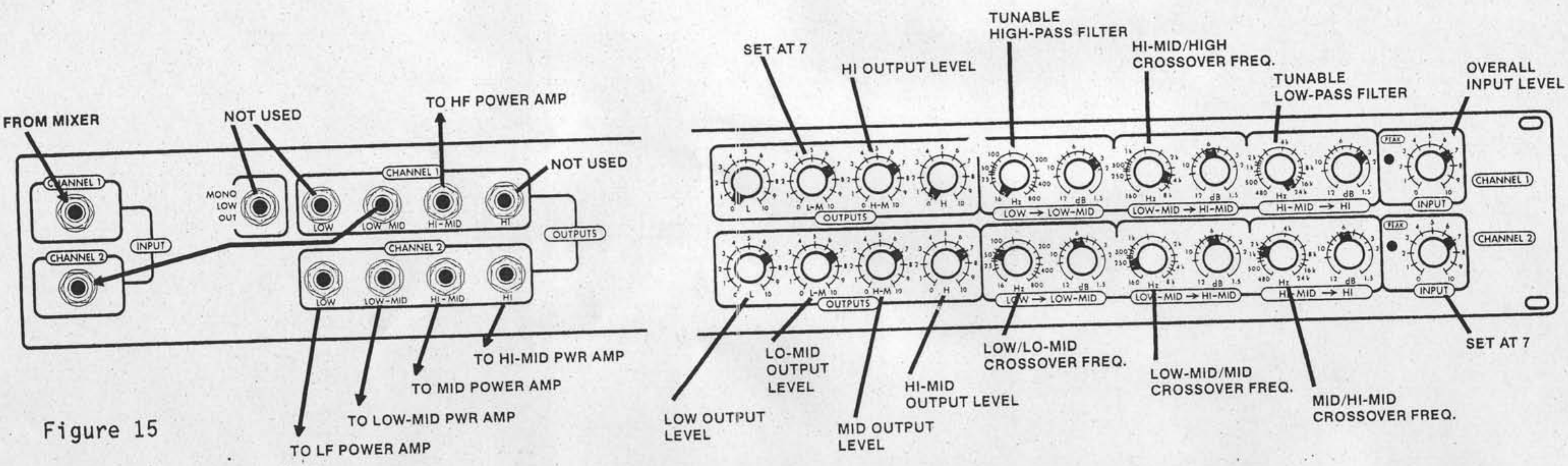


Figure 15

SC-88 USED AS A MONO 6-WAY WITH TUNABLE SUBSONIC FILTER

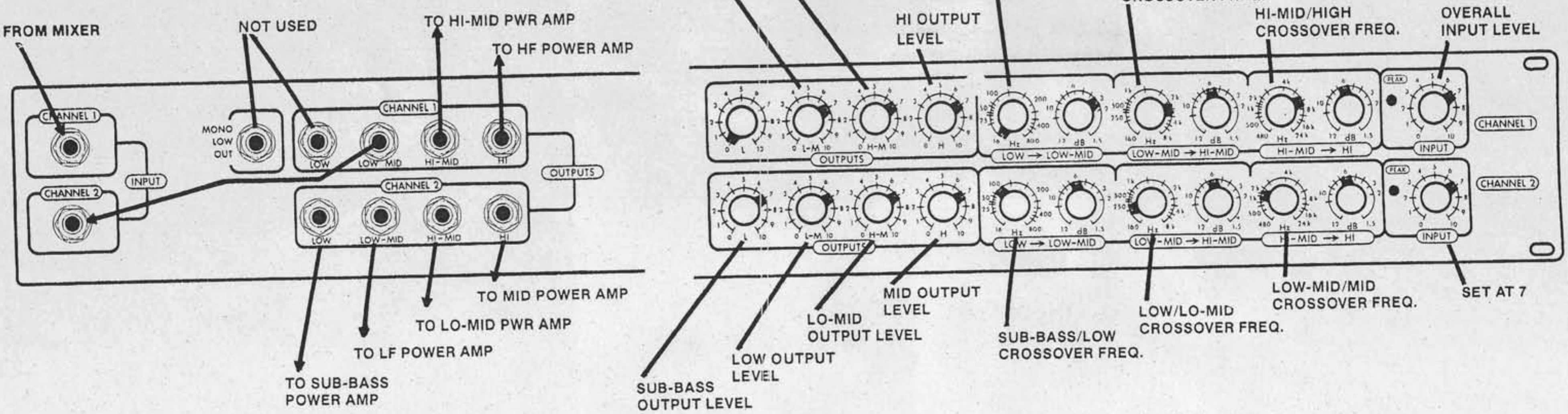


Figure 16

SC-88 USED AS A MONO 6-WAY WITH TUNABLE ULTRASONIC FILTER

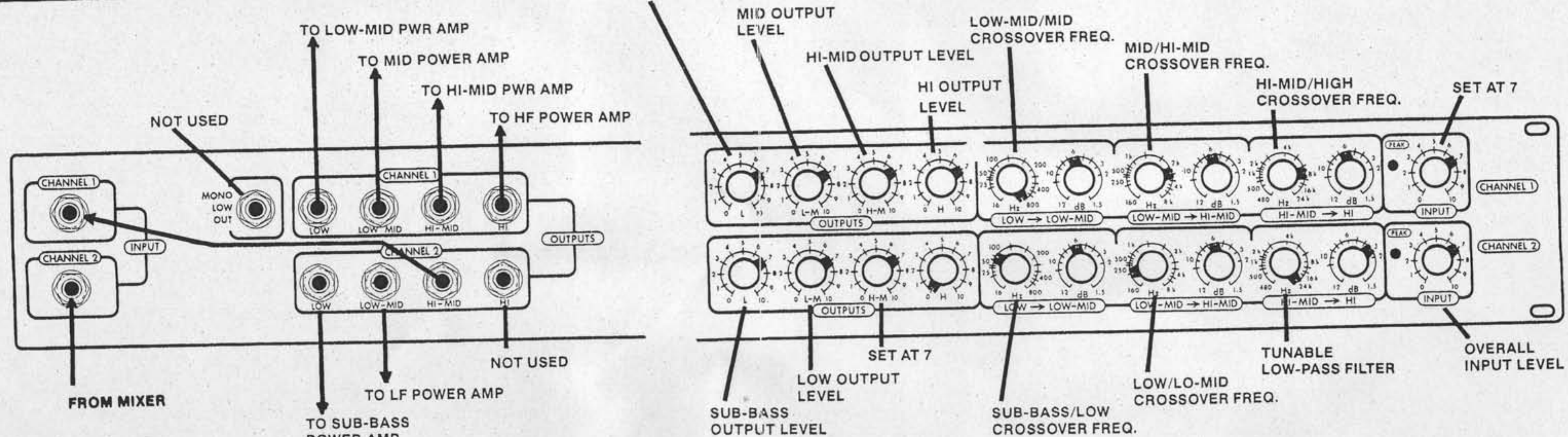


Figure 17

SC-88 USED AS A MONO 7-WAY

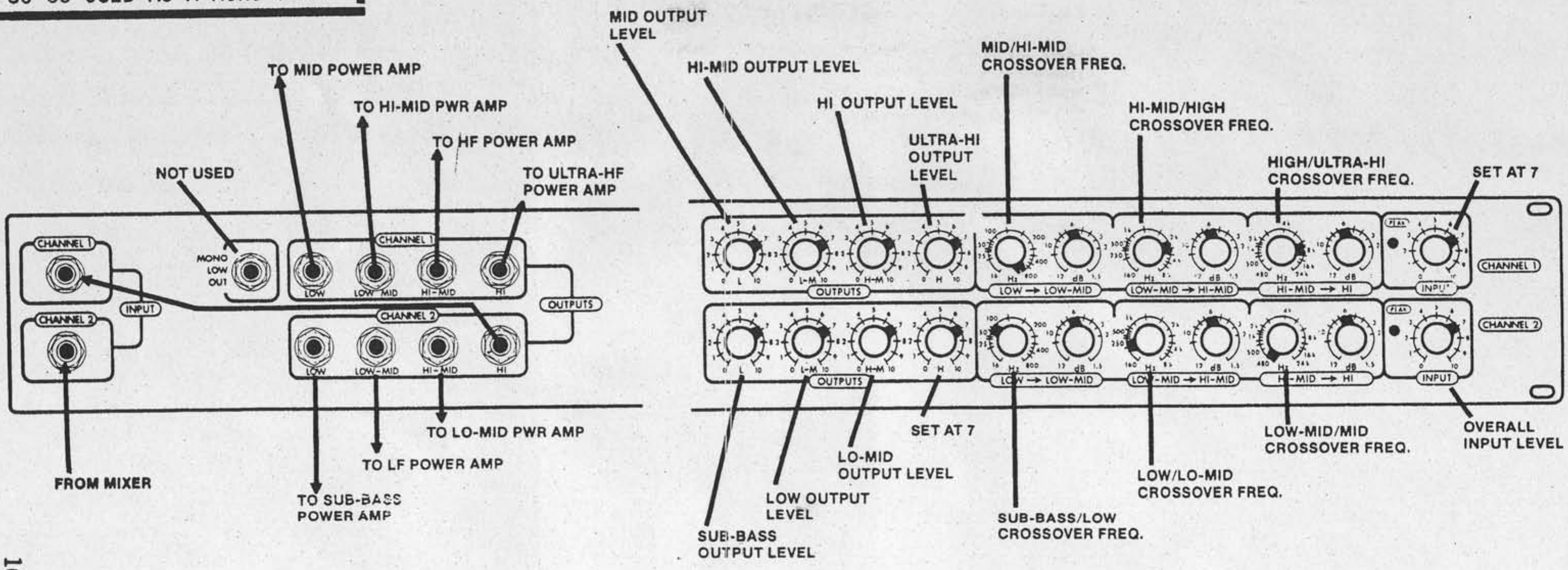


Figure 18

A CLOSER LOOK AT CROSSOVERS

INTRODUCTION

The bulk of this instruction manual is concerned with helping a sound system operator to get his system set up and running, and to answer the questions "how do I plug it in" and "where should I set the front panel controls." For those who want more information about the design and application of crossovers, this section may be a good starting place, covering such topics as the need for crossovers, characteristics of filters used in audio, limitations of passive crossovers, solutions offered by biamplification, and more detailed information regarding correct speaker alignment. The information offered here is only an introduction to the subject, since an in-depth discussion would certainly fill dozens of books.

As you read this, please try to always keep in mind that your crossover is only one part of a sound system, and that the electrical performance of a crossover, although predictable and well behaved, will almost never be accurately reflected in the acoustic performance of a speaker array. Also, there is no "best" approach to reproducing music through loudspeakers; there are a variety of methods in popular use, and each method has attendant problems and benefits which make it suitable for a particular application.

WHY USE CROSSOVERS?

The ideal speaker system would use a single small loudspeaker to reproduce the entire audible sound spectrum at any desired power level. It would be inexpensive, easy to hook up, and have a well-controlled coverage pattern that would not vary significantly with frequency. Unfortunately, such a speaker does not exist. Perhaps the closest approximation we have is the single-driver headphone, which can reproduce most of the audio spectrum fairly accurately. When any larger amount of volume is required, it becomes necessary to move greater amounts of air and larger speakers are required. For a given loudness, reproduction of low frequencies will demand greater changes in air pressure than high frequencies, and so will require a greater surface area in contact with the air. A speaker capable of good low-frequency response, then, will be relatively large. This characteristic does not favor good high frequency response, however, since the sheer physical mass of the large speaker inhibits the rapid back-and-forth movement required to reproduce high frequencies. A speaker suitable for reproducing high frequencies should be light and small. This disparity between low and high frequency requirements gets worse as the frequency response extremes and power demanded of a system get larger.

If one speaker cannot satisfy all requirements, then the solution must be to use two speakers, each suited to a particular portion of the frequency spectrum, and let each do its own job. This neatly solves one problem but instantly creates several new ones. First of all, if the woofer and tweeter are both wired in parallel and hooked up to a high power amplifier, there's a good chance that the tweeter will be destroyed. The tweeter will attempt to reproduce low frequency information fed to it, but its cone will not allow the long back and forth excursions necessary for lower frequencies, and so the excess power being fed to the tweeter will be dissipated in its voice coil as heat, eventually ruining it. Secondly, if the woofer and tweeter are not well matched in terms of their sensitivity, then one range of frequencies will be too loud. Typically, high frequency speakers, particularly horn-loaded compression drivers, are much more efficient than low frequency speakers. Somehow, the system needs to be balanced for equal response across the audio spectrum.

One of the inherent advantages of the "ideal" single-speaker system is that the listener hears the entire frequency range radiating from a single "point-source". That is, all the audio information leaves the speaker from essentially the same place and at the same time, ensuring that the listener hears all frequencies in the correct time and spatial relation to each other. As soon as the single speaker approach is abandoned, this important advantage is lost. Now, the sound is coming from two or more speakers at two different distances from the listener, and timing and phase errors are the result. See figure 19.

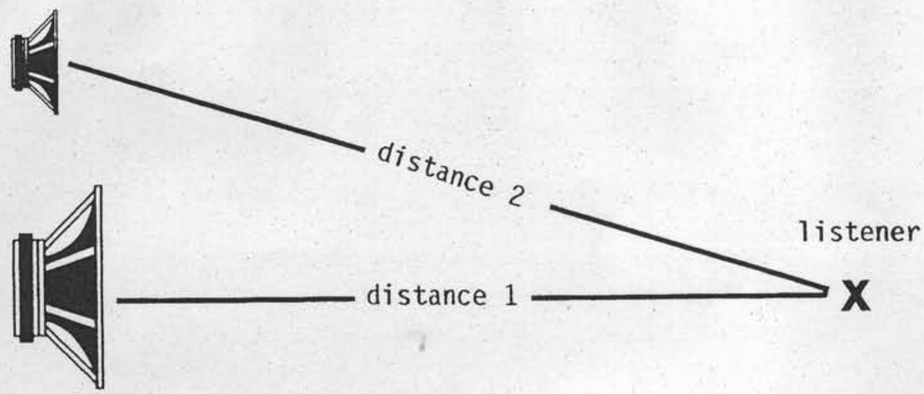


Figure 19 A listener is unlikely to be equidistant from two speakers in a multi-speaker setup, resulting in time and phase inaccuracies at the listening location.

The resulting errors are most pronounced in that range of frequencies where both speakers are contributing equal volume. In that case, you have the same audio information radiating from two discrete sources at the same time. Ideally, the two resulting wavefronts should combine perfectly to reach the listener at the same time, pushing and pulling the air in perfect synchronization. What usually happens, however, is that the two wavefronts reach the listener at different times and out of sync (out of phase). If the two sound sources happen to be exactly 180^{o} out of phase, then the listener will experience a significant null (hole) in the sound over some range of frequencies. If the listener then changes his position, all of the time and phase relationships between himself and the speakers will change. These relationships vary with frequency, too. In general, the more speakers that are used in a system, the more errors that will result. This remains true in most current professional sound systems.

Although there aren't many good solutions to this latter problem of physical misalignment in a multispeaker system, there are, at least, solutions to the problems of tweeter protection and sensitivity matching. By employing frequency selective crossover networks, we can route low frequency material to woofers and high frequency material to tweeters. This is accomplished by the use of filters, either passive or active, which ensure that each speaker will only "hear" that range of frequencies which it is capable of reproducing.

with the frequency spectrum thus divided, it becomes possible to discard some of the signal being sent to the tweeter, making level-matching with the woofer a simple matter.

PASSIVE CROSSOVERS AND FULL RANGE SYSTEMS

The simplest type of crossover feeds the full-range amplifier output to a crossover network with two passive filters, a low-pass and a high-pass. The appropriate filter outputs are then connected to the woofer and tweeter, as shown in figure 20.

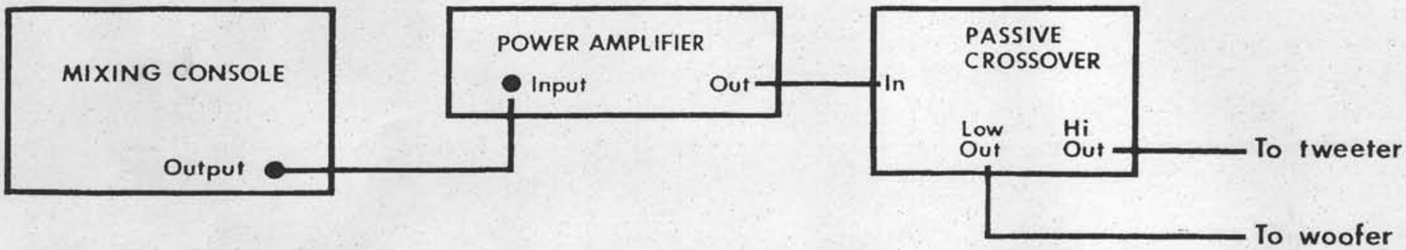


Figure 20 Passive Crossover

The advantages of the simple passive crossover are that they are simple to hook up, they accomplish their primary job of speaker protection, and they are cost-effective for low power applications. When built by a loudspeaker manufacturer for inclusion in a specific speaker system, the characteristics of the individual filters can be tailored to the requirements of the system.

However, the list of problems created by the use of passive crossovers is fairly lengthy, particularly when used in modern high-power PA systems. First of all, they offer no flexibility to the sound system operator. Passive crossovers have fixed operating parameters; you can't change their crossover frequency or filter shape. If your speaker system requirements change, you're stuck.

Other problems have to do with the components of the passive filters themselves, notably the inductors. Inductors are physically quite bulky, the more so for low frequency and high power applications, making them difficult and expensive to manufacture. Electrically, they are non-linear and subject to saturation and ringing, and are very sensitive to external magnetic fields.

Passive crossovers are difficult to design because they are always terminated by speakers, which present varying loads to the network. As the impedance of a speaker varies with frequency, so will the behavior of the filter to which it is connected. This interaction between speaker and crossover is of no help when you're trying to achieve a certain response from your system.

Using one amplifier for the entire audio spectrum is also found to be less than ideal. For one thing, it doesn't adequately answer the problem of mismatched sensitivities of the woofer and tweeter. If a horn loaded compression driver is much louder than the woofer in a passive system, the only answer is to "throw away" some of the high frequency signal by use of an attenuator on the crossover's high frequency output. This is certainly not efficient use of expensive amplifier power. Also, the greater the range of frequencies an amplifier is asked to handle, the more intermodulation distortion products you can expect to see.

All of these problems have been answered pretty well by the use of active crossovers and multi-amplified systems, but of course those alternatives present some new and unique problems as well. It remains true that, despite their limitations, passive crossovers still have value, as evidenced by their continued use in the majority of consumer loudspeakers and low-power PA and recording monitor applications.

ACTIVE CROSSOVERS IN MULTI-AMPLIFIED SYSTEMS

Active crossovers evolved as a solution to the previously mentioned problems of passive systems. In general, they are smaller, lighter, cheaper, more flexible, more efficient, and have less distortion than passive types.

In an active system, the audio is frequency divided at line level before being fed to the power amplifiers, as shown in figure 21 below.

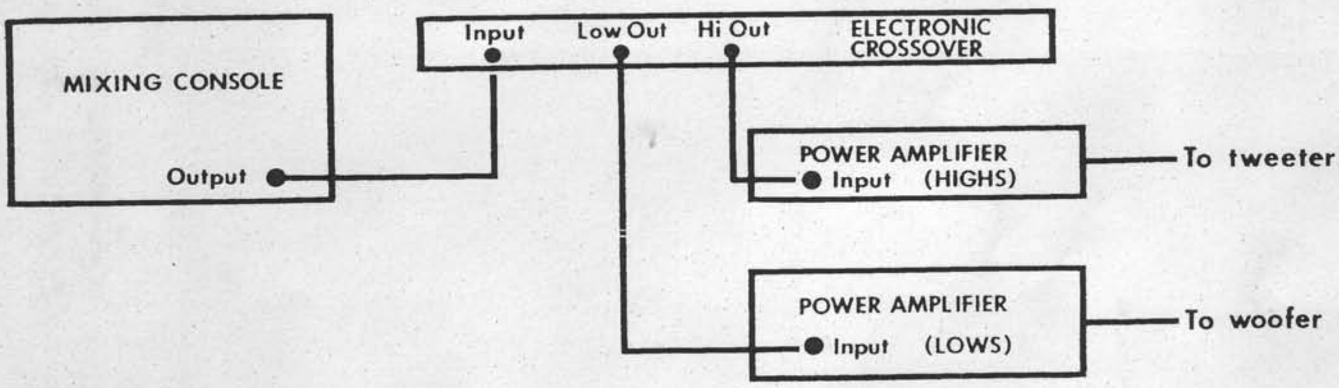


Figure 21 Biamplification

This system has a number of important advantages. The active filters in the electronic crossover eliminate the need for inductors, thereby reducing size, weight, cost, distortion, and hum. Active filters can easily be made tunable, which means that the crossover can be precisely modified to fit the speaker system at any time, a prime requirement for touring PA systems and fixed installations as well. The filters will respond in a stable and predictable way because they are terminated by a constant impedance within the crossover itself.

Since the amplifiers are asked to work within a more restricted bandwidth, IM distortion will be lower, and since each speaker will be driven by its own amplifier, amp power can be distributed efficiently within each frequency range. A given system may require a 250 watt amp for the woofer and only 40 watts for the horn.

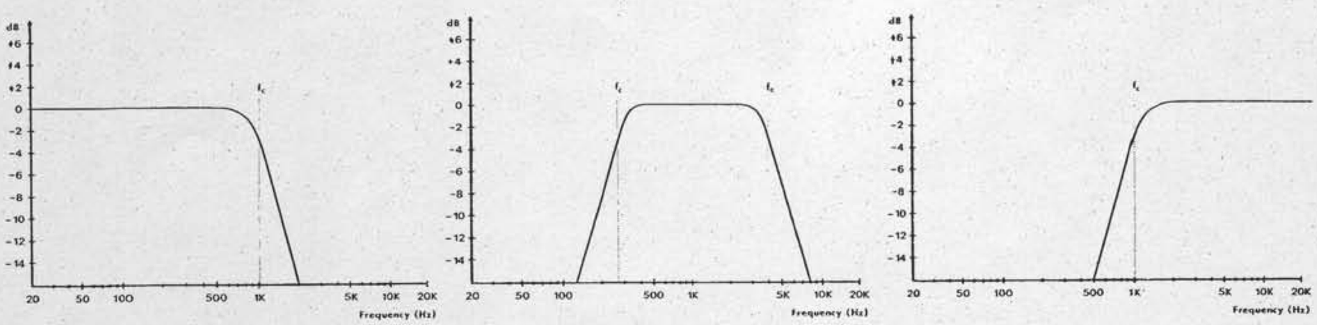
One of the nicest benefits of an active system is an increase in apparent loudness for a given amount of amplifier power. This is a result of the fact that low frequency amps can be overdriven without affecting the sound of the high frequency drivers. Low frequency information requires significantly more power from a system than high-frequency audio, and typically it is this low frequency audio that drives a system into distortion. If only one amp is powering a system, as in a passive or full-range setup, then important midrange and high frequency audio will be distorted every time a deep bass note overloads the amplifier. In a multi-amplified system, a low-frequency amp can overload without affecting higher frequency audio in any way. But that's not all. Fortunately, our hearing is less sensitive to distortion of low frequencies than midrange. Large multi-amped systems are frequently operated with what would be considered "unthinkable" levels of distortion in the lower bass region, and yet the system may sound very clean and loud to even a knowledgeable listener.

Some problems remain. The initial cost and complexity of a multi-amplified system are usually greater than passive systems, but it can be reasonably argued that the investment pays for itself in better sound per dollar.

FILTERS

Whether your crossover is a passive, high-level type, or an active, low-level type, it employs filters to accomplish its dividing job. Therefore, a quick look at the nature of filters seems relevant to an understanding of your crossover.

A filter is a frequency-selective electrical network which is designed to pass a certain range of frequencies while rejecting other frequencies. All crossovers are made with filters, but not all filters are identical. Filters may have a variety of characteristics and these are chosen to suit a particular audio requirement. The most common types of filters used in crossovers are low-pass, band-pass, and high-pass filters, supplying low-frequency, midrange, and high-frequency outputs respectively. A low-pass filter, as the name implies, passes low frequencies, up to a certain maximum frequency. Above this cutoff frequency, the signal will be attenuated (rejected) to some degree. A bandpass filter will pass a certain median band of frequencies while attenuating any frequencies above or below those desired. A high-pass filter performs the opposite function of the low-pass, passing all frequencies above a certain cutoff frequency while attenuating those below. These basic filter characteristics are usually shown in graph form as frequency vs. amplitude, as in figure 22 below.



(a) low-pass (b) bandpass (c) high-pass

Figure 22 Basic filter characteristics

Beyond these basic classifications of filter types, the audio designer is concerned with the details of filter performance near and within the cutoff or stopband. In other words, we'll assume that a filter is fairly linear within its pass-band (for example, the flat or "plateau" portion of the low-pass response shown in figure 22(a), but how does it behave as frequencies approach the cutoff frequency? And, once the filter is operating in its cutoff range, how quickly does it attenuate those undesired frequencies? Filters are not brick walls; they will always pass frequencies outside their pass band to some extent.

SLOPE

The rate at which a filter attenuates frequencies outside its passband is known as the slope, and is generally expressed in decibels of change per octave. For example, if a low-pass filter has a 12dB/octave slope, then any frequencies outside its pass band will be reduced in volume by an additional 12dB for each octave above the cutoff frequency. If the low-pass filter has a cutoff frequency of 1kHz, then a 2kHz signal can be expected to be reduced in volume by 12dB. Likewise, a 4kHz signal will be reduced by 24dB, and an 8kHz signal will be 36dB down.

The slope of a filter is a function of the number of frequency-reactive components within the filter. A very gentle slope, such as would be found in a 6dB/octave passive filter, might contain only one such component in the form of a capacitor connected from the audio to ground, as shown in figure 23. A capacitor passes high frequencies very well while blocking low frequencies. Therefore, any high frequency audio will be shorted to ground and not heard at the output. This type of filter is called a first-order filter. It is not terribly useful for crossovers because of its gentle slope; it won't provide much protection to high-frequency drivers when used in its high-pass version.

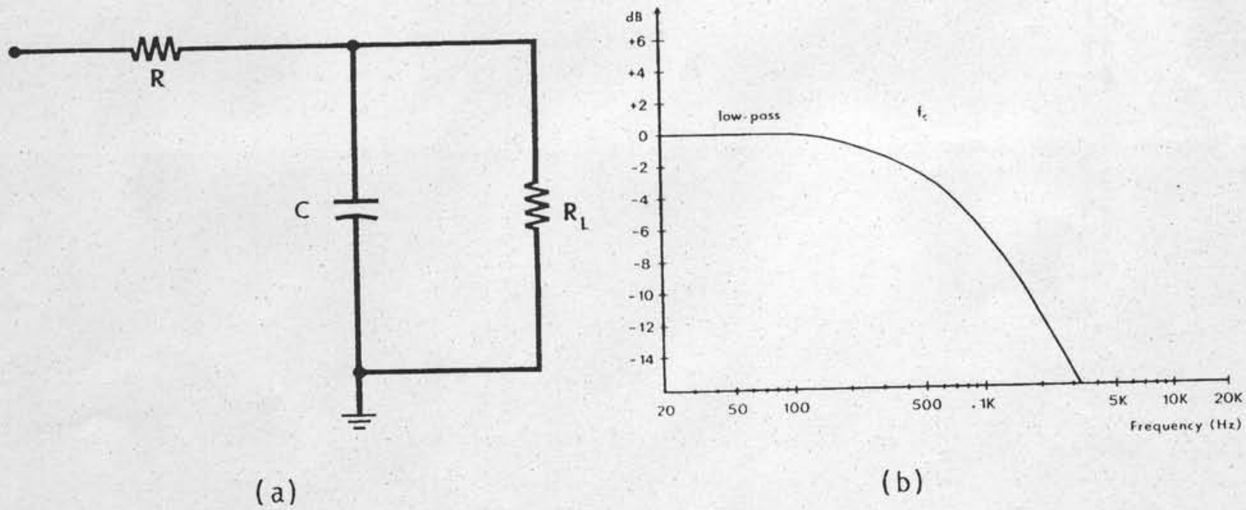


Figure 23 Passive RC first order low-pass filter and response.

High frequency loudspeakers will be better protected by a steeper slope, which can be obtained with a second-order filter such as the passive LC second-order filter shown in figure 24. Here, there are two frequency-reactive components: a capacitor (C), which easily passes high frequencies, and an inductor (L), which easily passes low frequencies. The slope will be 12dB/octave.

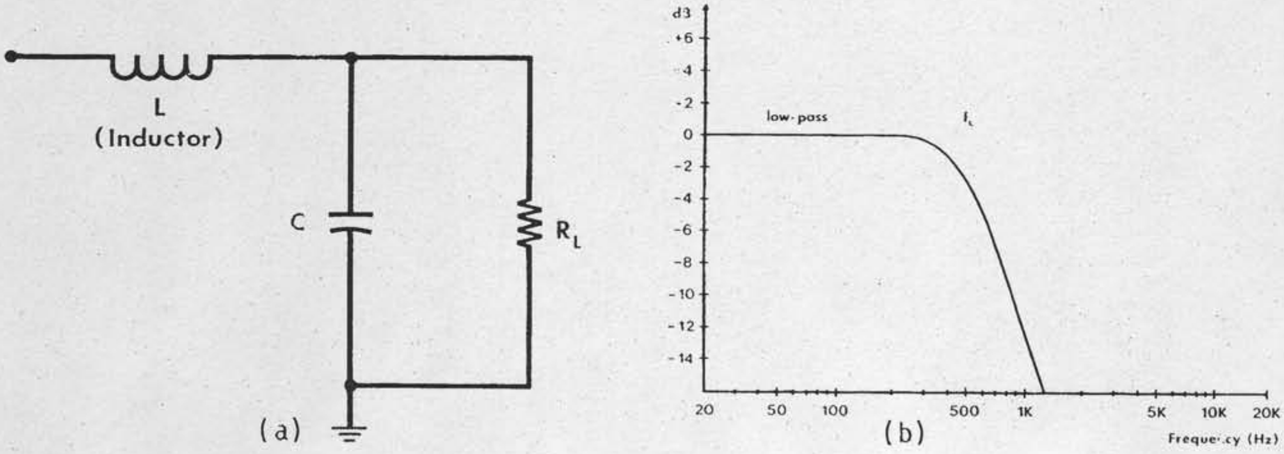


Figure 24 Passive LC 2nd order low-pass filter and response.

RESPONSE SHAPE

The performance of the filter in the immediate vicinity of the cutoff frequency is important to consider, as it will affect the way a filter sounds and the way in which its response combines with another filter in a multiple-filter system. Some filters begin to attenuate well before the cutoff frequency, while some remain essentially flat up to cutoff before abruptly beginning to attenuate. Some filters even exhibit an increase in amplitude as they approach the cutoff point, and then quickly reverse themselves and begin to attenuate.

These various response characteristics each has an application, and some of the most common ones have been given names, including Butterworth, Bessel, Chebyshev, Cauer, and Elliptical. The Butterworth response shape is very popular in conventional crossovers. Also called a "maximally flat" response, it stays quite flat within the pass-band and then falls off with a very linear slope. In actuality, it does not remain perfectly flat right up to the cutoff frequency, but begins to roll off a little earlier, so that the response is down 3dB at the cutoff frequency. A response plot of a low-pass 12dB/octave Butterworth filter is shown in figure 25.

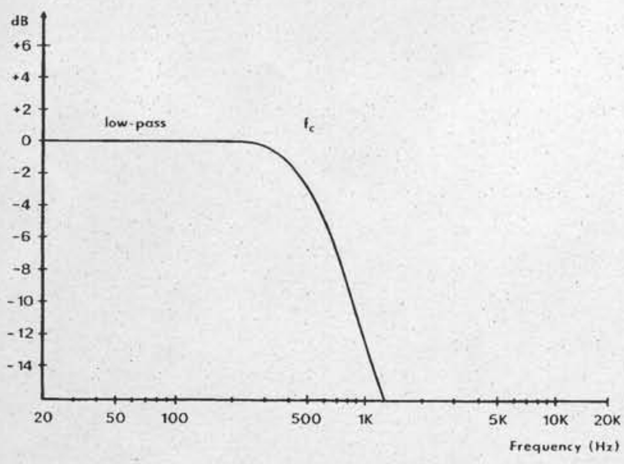


Figure 25 Butterworth response.

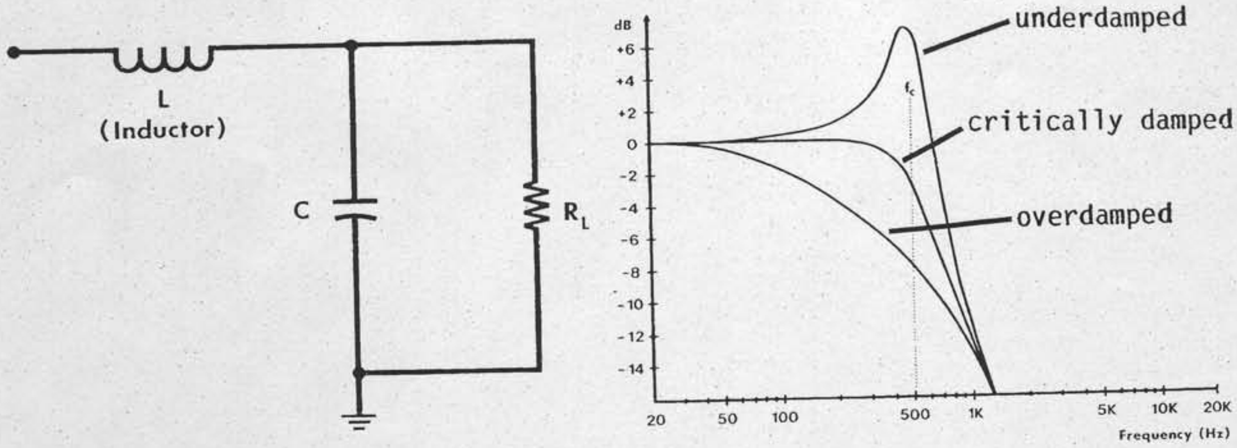
RESONANCE AND DAMPING

To understand the response characteristic of a particular filter, it is necessary to consider the factors of resonance and damping. This may be best accomplished by considering examples of resonance in everyday physical objects and then seeing equivalent effects in our filter circuits.

If you pluck a guitar string, it will vibrate back and forth at a certain frequency, called its resonant frequency. The vibrations will not go on indefinitely, since some part of your initial energy will be lost with each excursion. The energy will be dissipated by mechanical and acoustical friction, and also through the energy of the radiated sound. Eventually, the string will come to a halt. The process of energy absorption which slows and stops the string is called damping. Heavy damping, which would result if you touched the string with your finger, stifles the sound quickly, while a freely moving string would allow the tone to continue for a long period of time.

A loudspeaker system also has its own resonant characteristics, affected by the speaker itself, the cabinet it is mounted in, the damping material in the cabinet and even the room in which the speaker is located. When such a system is asked to reproduce frequencies which approach or coincide with its own resonant frequency, it begins to behave in a non-linear fashion. It offers much less resistance (impedance) to the stimulus, and as a result its output will be greater than at other frequencies even though the magnitude of the stimulus remained unchanged. The loudspeaker will be "boomy" or "peaky" at that frequency. However, if you stuff large amounts of absorbent fiberglass into the cabinet, that boominess will probably be attenuated somewhat, owing to the acoustical damping effect of the fiberglass on the cabinet's resonance. This example of mechanical damping has an electrical equivalent in the filters in your crossover. That is, a filter has a resonant frequency of its own, and can actually have a tendency to oscillate if so designed. On the other hand, the filter can be designed so that its resonance is "heavily damped," thus stifling any tendency to oscillate. This characteristic will be seen to be a very important factor in the summed response and overall sound of a crossover.

Let's take another look at a simple passive LC second-order low-pass filter, shown again in figure 26 below.



(a) (b)

Figure 26.

The filter of figure 26(a) is resonant at the cutoff frequency, and that frequency is proportional to the product of the inductor and capacitor values, L and C. We can control not only the product of the inductor and the capacitor, but their ratio as well. For example, if the capacitor is made very large and the inductor very small, the load resistor, R_L will not significantly load the LC circuit. The circuit then behaves as a series-resonant circuit with relatively low losses, and will in fact be on the verge of oscillation. For frequencies near resonance, the circuit will exhibit voltage gain or peaking, much like our resonant speaker cabinet example. A common name for this type of circuit behavior is underdamped response, and its response curve is plotted in figure 26(b).

By balancing the ratio of R, C, and L in the same second-order circuit, we can produce a very flat response curve with no peaking at all near the cutoff frequency, f_C. This type of response is known as a critically damped curve. You may also hear it referred to as a maximally flat or Butterworth response.

To go a bit further, if we use a very small capacitor and a very large inductor, the load resistor dominates and gives a very droopy, highly damped, response as shown by the lower curve in figure 26(b).

Realize that all three of these response patterns (over, under, and critically damped) start out at unity gain and end up rolling off at the same ultimate slope, in this case 12dB/octave. Also, the cutoff frequency remains the same. Setting the damping by changing the inductor-capacitor ratio determines only the shape of the response curve near the cutoff frequency.

You might also hear damping referred to as "Q". Actually, Q is simply the inverse of damping. An underdamped, peaky response has a very high Q, while an overdamped response has a Q which tends toward zero. A critically damped filter has a Q of 0.5.

ACTIVE FILTERS

The filter examples given so far have used passive types for purposes of illustration. These were the first filters used for audio, and although their performance can be very good under some conditions, they have been almost universally replaced in modern electronic crossovers by active filters. By combining resistors and capacitors with IC op-amps, we can accurately simulate the performance of traditional inductance-capacitance filters.

The advantages of active filters are many. They are inexpensive, lightweight, and compact, the more so at very low frequencies, where inductors would be large, heavy, and expensive. They are easily tuned over a wide range of frequencies, they are largely unaffected by external loading, and they don't require exotic shielding to protect them from magnetically induced hum.

Having now touched upon some basic filter characteristics, we can get back to the main subject of discussion, namely loudspeaker crossover networks.

HOW CROSSOVERS ARE BUILT

The majority of the commercially available electronic crossovers are made by simply combining an active low-pass filter and an active high-pass filter. Usually, the response shape of each filter is Butterworth, and a single front panel control tunes the cutoff frequency of both filters simultaneously. This type of crossover is illustrated in figure 27.

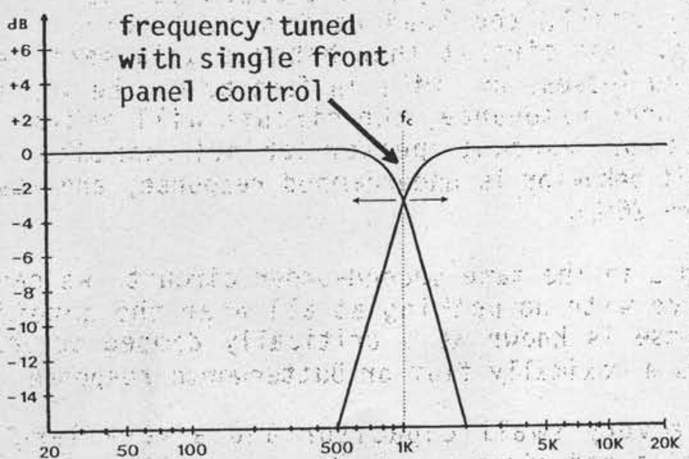
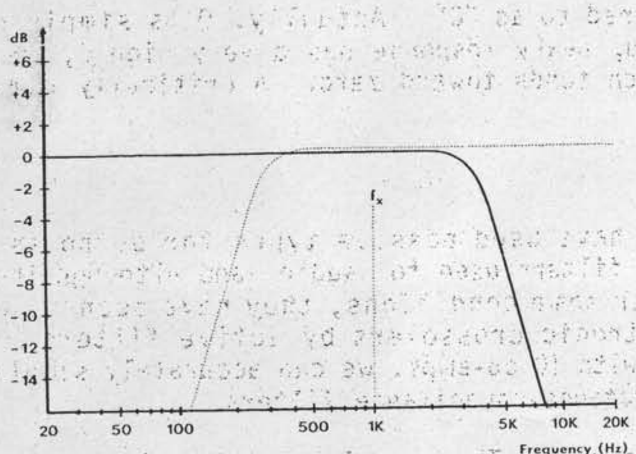


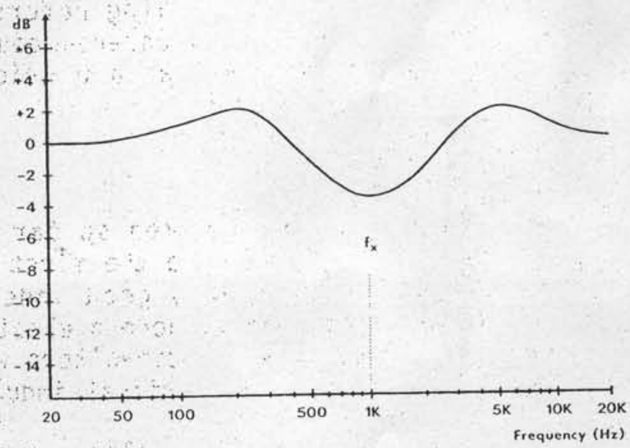
Figure 27.

A typical Butterworth-response 2-way electronic crossover, made by combining a low-pass and high-pass filter. Note that the crossover frequency is at the "3dB-down" point of each filter. This approach to crossover design works fairly well, but the response shape of each filter is fixed, and therefore there can be no control over the summed response of the network at the crossover frequency. The summed response will show a +3dB "bump" at the crossover frequency.

Another popular approach uses the same two overlapped filters just shown, but gives the operator independent control of each filter's cutoff frequency, ie., two front-panel frequency adjustments for each crossover point. This provides an opportunity to either overlap the two pass bands or pull them apart. The result may not be quite what you had in mind, as illustrated in figure 28.



(a)



(b)

Figure 28.

Here, the low-pass and high-pass filters are shown overlapping by two octaves in each direction past the crossover point. You might logically expect a nice even boost in summed response throughout this overlap region, but that's not what you'll actually get.

The two filters of (a), when electronically summed, will combine to give you a very uneven response curve with two peaks near the individual cutoff frequencies, and a dip in the crossover region.

Ashly electronic crossovers, unlike the two examples just given, do not use two fixed-response filters to form a crossover. Instead, a single state-variable filter provides both low-pass and high-pass outputs, and a single control adjusts the crossover frequency. Because the state-variable filter allows complete control over all filter parameters, it is possible to tune the response shape of the filter for the best possible summed response. As you'll recall, response shape is determined by filter damping, and so all Ashly crossovers feature a damping control, labeled dB, at each crossover point.

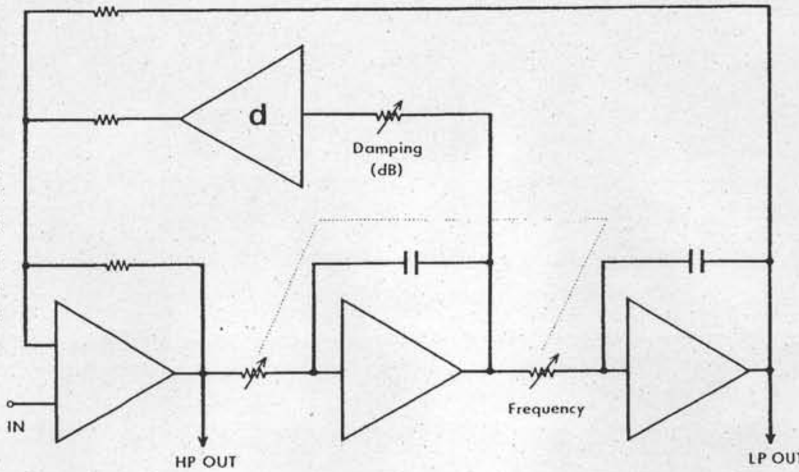
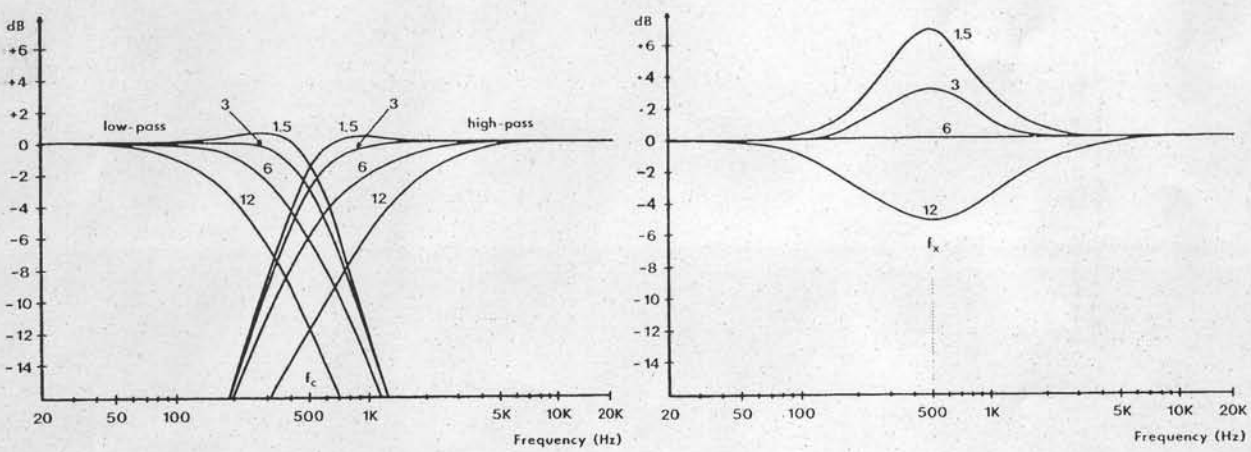


Figure 29

This state-variable filter, representative of the filters used in Ashly SC-series crossovers, is an analog computer realization of a second-order (two pole or two integrator) filter system.

From the standpoint of the sound system operator, the Ashly filter approach combines the one-knob frequency control convenience of the first crossover example (fig. 27) with the summed response flexibility of the second crossover example (fig. 28). In contrast to the bumpy response of fig. 28, Ashly crossovers exhibit the smooth and predictable summed response curves shown below.



(a) Figure 30. (b)

This graph plots the high-pass and low-pass outputs of the Ashly 12dB/octave filter for 4 different settings of the dB control. The line marked "3" shows Butterworth response, and is obtained by setting the control to 3. The other curves show the response shapes obtained by over and underdamping the filter.

This graph shows how the response curves of (a) combine when electronically summed. The Butterworth curves sum with a gentle 3dB peak, but other responses are easily obtained; for a flat summed response, set the dB control to 6. (12dB per octave models only)

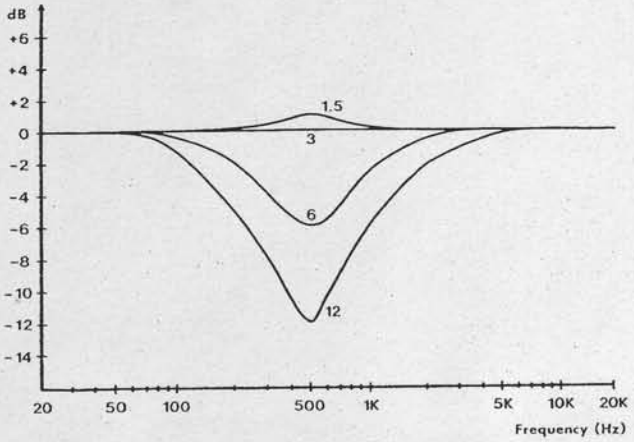


Figure 31

The summed amplitude response of the Ashly 18dB per octave crossovers. For these models, flat summing is achieved by setting the dB control to 3.

Another nice feature of the state-variable filter is that once you set the damping for a particular response, you don't need to re-adjust it for new crossover frequencies. For example, if you own an Ashly 12dB per octave crossover and have set the damping (dB) control to 6 for flat summing, you can then set the crossover frequency anywhere you want and the outputs will always sum flat. This is not true of other crossover designs.

At this point, you might ask why we don't simply tune the filter responses for flat summing and remove the rolloff (dB) control from the front panel altogether. After all, who wants less than a perfectly flat response? To answer that question, you have to put everything back into perspective and remember that the crossover is not going to be connected to a resistive summing network. In the real world, you'll be hooking the crossover up to amplifiers which will in turn be connected to speakers which will in turn be acoustically coupled to the air and to each other. Loudspeaker reponse is not likely to be ruler flat at the crossover frequency; instead, there will likely be a hot spot or a deficiency in response due to the imperfect nature of the speakers. When that happens, all those careful calculations about flat summing fall flat on their face. All of the sudden, you find that you need an equalizer right at the crossover point, and with an Ashly crossover, you've already got it. Take another look at those response curves of figures 30 & 31 above; in practive, being able to modify the damping of the filter in your crossover is just like having an equalizer tuned precisely to your chosen crossover frequency. The Ashly SC series crossovers give you the best of both worlds: electrically flat summing to satisfy the lab technician, and a choice of response shapes to help you approach acoustically flat summing where it really counts--in a real-life loudspeaker system.

CROSSOVER EVALUATION

A popular and convenient method of evaluating a particular crossover is to look at summed amplitude response, which can be accomplished by simply wiring all the outputs of the crossover together in parallel through an appropriate resistive network. The theory behind this sort of examination is that if you add all the outputs together, the sum of the individual pass-bands should equal the input signal. That is, the crossover should be capable of a flat summed response. It is often said, incorrectly, that 12dB per octave crossovers are incapable of flat summed response, while 18dB per octave types do sum flat. The two determining factors in summed response, response shape and phase relationship between filters, are frequently ignored or misunderstood.

PHASE SHIFT

When audio is passed through a filter, there will be a shift in the phase of the signal. This is true of both active and passive filters. For example, in a simple second-order (12dB/octave) crossover network, the low-pass output will lag the high pass output by 180^{0}. That means that there will be an actual time difference in the movement of woofer and tweeter at the crossover frequency; as the tweeter moves forward, the woofer will be moving backwards, since the woofer will always be 1/2 wavelength behind the tweeter. If the two drivers are of equal efficiency at the crossover frequency, their simultaneous pushing and pulling will cancel each other out, and the result will be an audible notch at the crossover frequency. This is even more obvious when the two filters are electrically summed--there will be a very deep notch in the summed response curve where the two filters meet and cancel each other out. Fortunately, there is an easy solution.

When two signals are exactly 180^{0} out of phase, it is a simple matter to invert the polarity of one output, thus eliminating the notch effect. That explains why you'll often hear people say the the outputs of a 12dB per octave crossover should be wired out of phase. This is one way of getting around the phase error inherent in these crossovers. However, this would be a mistake with an Ashly crossover; we have already performed all necessary phase inversions internally in our crossovers so that the user can simply wire everything in phase without having to stop and think about which outputs to invert and which ones not to invert. Unfortunately, this is not an agreed-upon standard among crossover manufacturers, and so there is confusion on this point.

18dB per octave crossovers also have filter phase shifts, but unlike the second-order crossovers, their outputs come out 90^{0} out of phase, a difference of 1/4 wavelength. When electrically or acoustically summed, these outputs will not produce the notches associated with second-order crossovers. Phase inversion and polarity swapping are not necessary.

RESPONSE SHAPE

Another source of confusion in summed amplitude response testing is response shape of the individual filters. You'll hear it said that, even when wired in-phase, 12dB per octave crossovers won't sum flat. It is true that 12dB per octave **Butterworth** shape filters won't sum flat, and this is the source of the unfortunate generalization. However, if the filters are more highly damped, then flat summing is certainly possible. This was illustrated in figure 30.

CHOICE OF SLOPE

Once it is realized that the slope of a crossover doesn't necessarily have anything to do with summed amplitude response, we can proceed to choose a crossover slope based on more important criteria. The choice of a particular slope is largely subjective; some people like the "sound" of an 18dB per octave crossover, while others might feel that its steep cutoff characteristic makes it sound harsh. One thing that is clear, though, is that a steeper slope will afford more protection to high frequency drivers than a gentle rolloff. For the majority of audio applications, a 12dB per octave crossover will be found to be perfectly adequate. Where extra protection is deemed necessary, choose the 18dB per octave type.

SUMMED PHASE RESPONSE

Nearly all crossovers introduce phase shifts that vary as a function of frequency, but this is not considered to be a problem since our ears don't seem to be very sensitive to phase shift. Just to set up a suitable A/B comparison to test the audibility of phase shift would be quite a feat, since it would require near-perfect speakers and source material. After all, phase shifts are introduced virtually everywhere in an audio system; microphones, transformers, equalizers of all kinds, effects units, crossovers and speakers all contribute significant amounts of phase error. In short, crossover phase shift effects are not considered to be an important factor in the accuracy of a sound system.

SETTING UP YOUR SYSTEM FOR ACCURATE REPRODUCTION

In order to realize the best possible performance from your sound system, it is desirable to strive for the best possible phase relationships between separate drivers. Since it is impossible to have all of your speakers radiate their energy from the same point in space, your only option is try to keep their acoustical centers coherent in at least one plane, i.e., try to stack the speakers with the drivers centered in one vertical line. This will minimize phase cancellations and improve the projection pattern of the system. Notice we aren't saying you'll achieve perfect phase correlation in your system, because you won't. There are simply too many variables in a multi-speaker system.

For example, we've recommended that you stack your speakers in one vertical plane, but what if you've got 2, 4, or 6 bass bottoms per side, all reproducing the same frequency? In that case, you've got wavefronts radiating from physically separate points in both the vertical and horizontal plane, which means that the system's polar pattern and amplitude response will be subject to the way those wavefronts add and subtract as they recombine.

Then, there's the knotty problem of determining where the wavefront actually emanates from in a real speaker. Does it radiate from the diaphram in the driver, and should that be considered the acoustical source for alignment purposes? Or, if that driver is connected to an exponential horn, should we consider the mouth of the horn the correct alignment point? What's the phase relationship between the mouth of the horn and the diaphram? What about a 15 inch woofer flush-mounted to a ported enclosure? Or a dual 15 inch folded "W" cabinet--where is its "acoustical center"? As you can see, there's room for error in a multi-speaker system. Generally, the more complex the system, the more difficult the phasing question.

All of this is not to say that contemporary sound systems don't work or sound good--they can and do. Just keep in mind that there are physical limitations in a real sound system that make perfect phase alignment nearly impossible. Again, if you'll try to keep the system aligned in at least one plane, you'll probably be doing the best you can.

SUMMARY

The ultimate goal of a sound system is the faithful reproduction of music and speech without additional coloration. Having departed from the ideal single-speaker approch, we now have a number of variables which will influence the overall sound, including the crossover, amplifiers, time and phase errors which result from having spatially displaced drivers, and the individual personalities of various kinds of speakers reproducing overlapping frequencies; an aluminum diaphram compression driver coupled to a fiberglass horn will sound different than a paper cone woofer, even though both may be reproducing the same frequency. Also, each speaker can be expected to have significant variations in its frequency response, even within its flattest range. The summed acoustic response of a loudspeaker system therefore becomes a function of the crossover, the amplifiers, the loudspeakers, and the room in which it's all used. Of these, the loudspeakers and room acoustics remain, by far, the greatest sources of error in typical installations.

We have given rough guidelines for setting up multiple loudspeakers, but sometimes even the most minimal attempts at speaker alignment will be difficult to implement. For example, in a touring sound system, first consideration may have to be given to setting up the speaker stack in such a way that it doesn't fall over or get in the way of a lighting truss, rather than to optimum phase alignment. It's worthwhile to keep a sense of humor when theorizing about the "perfect" speaker installation; the variables can be overwhelming.

There is no "best" approach to sound reproduction, since every application demands something unique from a system. A two-way passive system can sound great, and a 7-way multiamplified system can also sound great. It's probably true, however, that an unskilled operator can create more havoc with a complex system than with a simple one, so the value of simplicity should not be dismissed.

Just as there is no single best approach to sound reinforcement, there is no single best crossover characteristic which is best for all applications. A parametric-type crossover, offering control of crossover frequency, filter damping, and output levels, ensures enough flexibility to meet changing system requirements.

DEFINITION OF TERMS AS USED IN THIS MANUAL

ACTIVE

Electronic circuits which use devices such as transistors and integrated circuits, and which are capable of voltage and power gain as well as loss. Circuits using only resistors, capacitors, transformers, etc., are referred to as passive.

AMPLITUDE

The voltage level of a signal. May be measured in volts or decibels. Generally corresponds to the volume or intensity of an audio signal.

BALANCED

A 3-wire circuit arrangement in which two conductors are designated as signal lines (+ and -), and the third is a shield and chassis ground. The signal lines are of opposite polarity at any given moment, and are of equal potential with respect to ground. Balanced input amplifiers are used on all Ashly SC series products to improve hum and noise rejection. Jumpering signal minus (-) to ground provides an unbalanced input.

BUTTERWORTH

The name of a particular filter response shape. The response is essentially "flat" within the pass-band, is 3dB down at the cutoff frequency, and continues to attenuate at a constant slope. Also called a "maximally flat" or "critically damped" filter shape, it is very popular for crossovers and shelving filters.

CENTER FREQUENCY

The frequency (or pitch) at which a filter is most effective. In a parametric equalizer, it refers to the frequency where a particular boost/cut control has maximum effect.

DAMPING

A force which opposes the tendency of a system to oscillate.

dB

A unit by which audio levels can be COMPARED. Often thoroughly misunderstood are the concepts that decibels represent the level of a signal compared to some reference level (15 dB cut means a certain level less than a previous level --- the absolute level of the signal need not be known), and that decibels are a logarithmic unit.

Some handy numbers to remember when dealing with decibels:

+3 dB = Double Power
+6 dB = Double Amplitude, Quadruple Power
+10 dB = 10X Power
+20 dB = 10X Amplitude, 100X Power

dBm

A unit of measurement in decibels where 0 dBm = a power level of 1 milliwatt into a 600 ohm load. Originally defined by the telephone company to measure line levels.

dBV

Decibel Volts, an update of the dBm definition where 0 dBV = the same voltage level as 0 dBm, but with no regard to power or impedance. 0 dBV = 0.778 Volts. This unit is much more appropriate for modern audio equipment with high impedance inputs and low impedance outputs.

DISTORTION

Generally refers to ANY modification of an audio signal which produces new frequencies which were not in the original. Examples are harmonic distortion, where a circuit adds overtones to a fundamental signal, and intermodulation or IM distortion, where two frequencies beat together to produce sum and difference frequencies.

FEEDBACK

Generally refers to any process where an output is in some form routed back to an input to establish a loop. Negative feedback tends to be be self stabilizing, while positive feedback causes instability.

FILTER

A circuit designed to pass some frequencies, but not others. There are three general categories of filters: High-pass, band-pass, and low-pass. The high-pass filter passes frequencies above a certain limit, the low-pass passes frequencies below a limit, and the band-pass passes one group of frequencies without passing those above or below. Our equalizer uses band pass filters, crossovers use high and low-pass filters.

FREQUENCY

The repetition rate of a waveform. Frequency is measured in Hertz. One cycle per second (cps) is one Hertz (Hz). The higher a note on a musical scale the higher its frequency.

FREQUENCY RESPONSE

Refers to relative gain and loss at various frequencies across the audio band. May be illustrated by a graph called a frequency response plot, usually graphing decibels vs. Hertz or octaves.

HERTZ (Hz)

The unit of frequency measurement. (Formerly called Cycles-per-Second: this explains it perfectly)

HEADROOM

Refers to the increase in level above normal operating level that can be obtained without clipping. Usually expressed in dB.

IMPEDANCE

Essentially the AC equivalent of resistance. It describes the drive capability of an output, or the amount of drive required for an input at any given signal level.

KHz

Kilohertz. 1,000 Hertz.

LEVEL

The magnitude of a signal, expressed in decibels or volts.

LINE LEVEL
Meaning "somewhere around 0dBV" as opposed to MIC level of around -40dBV.

OCTAVE
A logarithmic unit to compare frequencies. +1 Octave means double frequency, -1 Octave means half frequency.

OHM
The unit of electrical resistance or impedance.

ORDER
A term describing the slope of a filter. A first order filter will have a slope of 6dB/octave, second-order will be 12dB/octve, third-order will be 18dB/octave, fourth-order will be 24dB/octave, and so on. Higher order filters are typically made by cascading lower-order filter sections.

PHASE
Describes how well two signals are in step. In-phase means that positive and negative peaks in two signals occur together, while out-of-phase means they do not occur together. Variations in signal timing as well as polarity can make two signals in or out of phase, or anywhere in between. Phase is usually measured in degrees where 0 degrees is in-phase, 180 degrees is out-of-phase, and 90 degrees is in between (sometimes called quadrature).

PREAMPLIFIER
The first stage of amplification, designed to boost very low level signals to line level.

"Q"
A measurement describing the sharpness or broadness of a filter.

RESONANCE
The tendency of an electrical or mechanical system to vibrate (or oscillate) at a certain frequency.

SHELVING
Describes an equalization action where all frequencies above or below a particular frequency are boost or cut.

SLOPE
In a filter or equalizer, a description of the rate of boost or attenuation. Normally specified in dB/octave (6, 12, 18, or 24dB/octave slopes are most common). The steeper the slope, the higher the "Q" in a filter.

TRANSIENT
A sudden burst of energy in an audio signal, such as a breath blast in a microphone, the sound of a snare drum, or a deep scratch in a record. Transients frequently reach peak levels of 10 to 30 dB above standard operating level, and may cause distortion or even damage to equipment.

UNITY GAIN

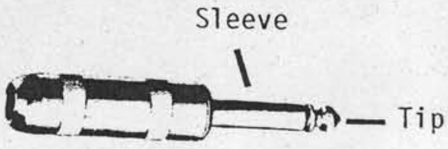
Output level = Input level.

WIRING, PHONE PLUG AND XLR

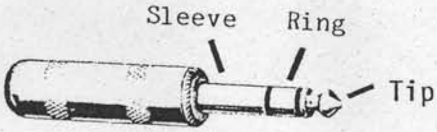
A stereo phone plug is wired + to the tip, - to the ring, and shield to the sleeve. For a mono phone plug, combine - and shield, and connect both to the sleeve.

An XLR (3 Pin) connector is wired + to pin 3, - to pin 2, and shield to pin 1.

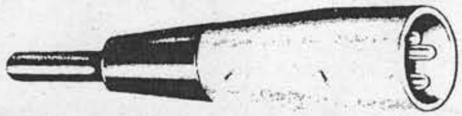
Mono Phone Plug:
(for unbalanced
inputs and outputs)



Stereo Phone Plug:
(for balanced in-
puts and outputs)



XLR Type Connector:
(Male Shown)



FREQUENTLY ASKED QUESTIONS ABOUT ASHLY CROSSOVERS

Q: Is the "dB" control a slope control? If I set it at 6, will my crossover have a slope of 6dB per octave?

A: No, this control has no effect on the ultimate slope of the crossover. Slope is a fixed parameter; i.e. a 12dB per octave model will always have an ultimate slope of 12dB per octave, regardless of front panel control settings. The rolloff (dB) control actually affects the damping of the filter circuit in the vicinity of the crossover frequency. In practice, this control acts much like an equalizer tuned to the crossover frequency, with clockwise rotation of the control producing a boost in the summed response at the crossover point, and a counterclockwise rotation producing a dip in the summed response. See pages 5 and 27.

Q: Where should I set the "dB" control?

A: We recommend an initial setting of 6 for our 12dB per octave models and a setting of 3 for our 18dB per octave models. Then, tune for flattest response or best sound.

Q: Are Ashly crossovers capable of flat summed amplitude response?

A: Yes. In addition, other summed response shapes are available via the "dB" control, helping to compensate for uneven speaker response.

Q: Can my crossover be connected to long snakes and drive low impedance loads?

A: Yes. The output stage is designed to tolerate low impedance and reactive loads without degrading performance, and provides good immunity from RF and other common-mode noise.

Q: What does a flashing peak light indicate?

A: You are either very close to or already in clipping. Our peak circuit monitors signal levels at several key points in the audio chain. See page 6.

Q: How should I choose my crossover frequencies?

A: Follow speaker manufacturers recommendations for lowest and highest crossover points. Too low of a crossover frequency may damage some high-frequency drivers. Within a given safe range of crossover frequencies, the actual choice of the crossover point is largely a matter of personal taste and summed system response.

Q: Should I wire any of the crossover outputs out of phase with each other?

A: No. All necessary phase inversions are done internally in Ashly crossovers.

TROUBLE SHOOTING TIPS

NO OUTPUT

Check AC power - is the pilot light on? Check in/out connections, are they reversed? Are you sure you have an input signal?

PEAK LIGHT FLASHES OR STAYS ON ALL THE TIME

If the LED continues to flash when all of the crossover level controls are turned down all the way, it indicates that the crossover is being fed excessively hot levels from a previous piece of equipment. Turn down your mixer. See page 6.

DISTORTED SOUND

Is the peak light flashing? If it is, an overload is occuring within the crossover, and may also be occurring in other parts of the system. If the peak light is not flashing, the distortion is occurring somewhere in the system prior to the crossover.

EXCESSIVE HUM OR NOISE

Hum will usually be caused by a "ground loop" between components. Try using the suggested balanced input and output hook-ups if the other pieces of equipment used in conjunction with your equalizer have balanced inputs and outputs. Noise can be caused by insufficient drive signal.

NOTE:

UN-SHIELDED CABLES, IMPROPERLY WIRED CONNECTIONS, AND CABLE WITH BROKEN STRANDS (SHORTS ETC.) ARE THE MOST COMMON PROBLEMS. MAKE SURE YOU USE GOOD QUALITY CABLE.

WHEN IN DOUBT, GET IN TOUCH WITH YOUR ASHLY DEALER, OR CALL THE FACTORY DIRECT - (800)828-6308. In New York State dial (716)544-5191.

SPECIFICATIONS

INPUT GAIN - ∞ - +10dB

ROLLOFF: 1.5dBV-12dB (crossover point depth)

OUTPUT GAIN - ∞ - +20dB

INPUT IMPEDANCE 10k Ω balanced bridging

OUTPUT IMPEDANCE 50 Ω unbalanced-terminate with 600 Ω or more

MAX. IN-OUT LEVEL +20dBV

FREQUENCY RESPONSE ±.5dB 20Hz-20kHz (within passband)

DISTORTION < .05% THD, +10dBV 20Hz-20kHz

HUM AND NOISE -90dBV

POWER 120 VAC, 50-60Hz, 5W.

BLOCK DIAGRAMS

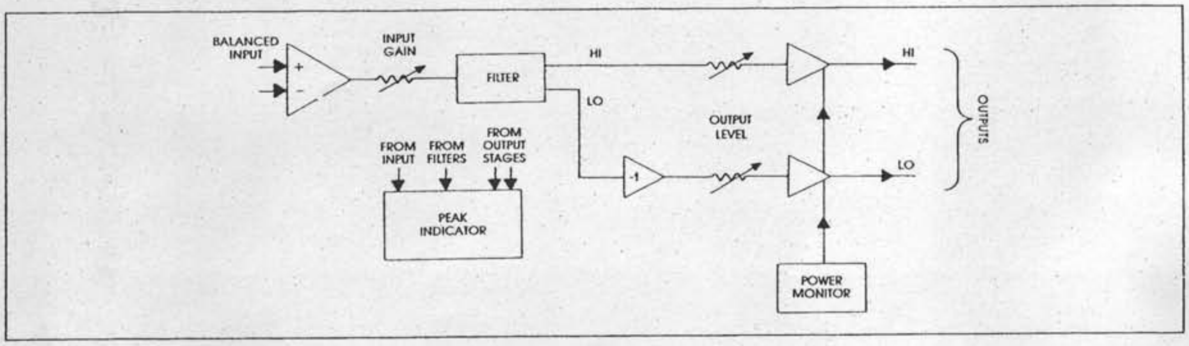


Figure 31 Block diagram of Ashly 2-way crossovers

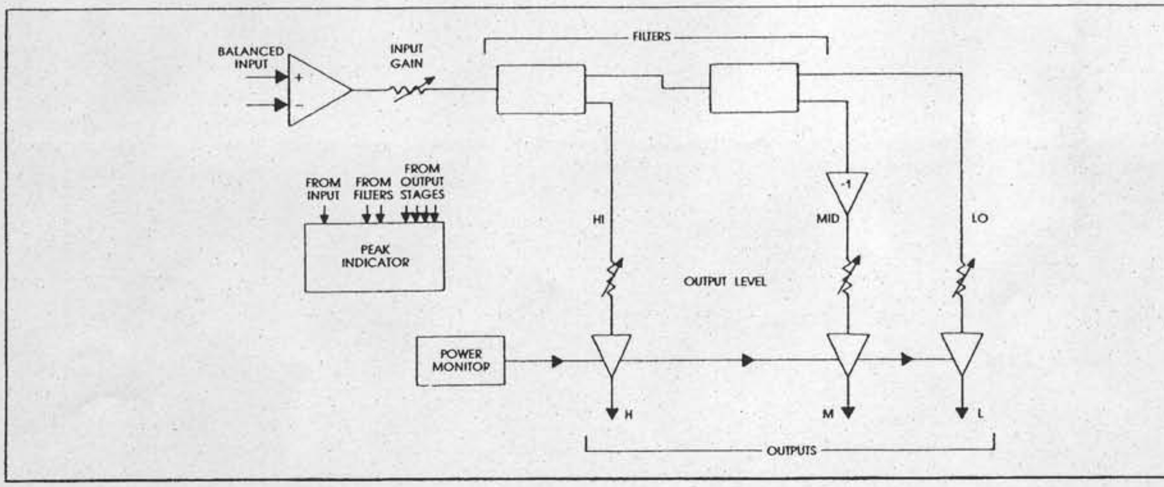


Figure 32 Block diagram of Ashly 3-way 12dB/octave crossovers

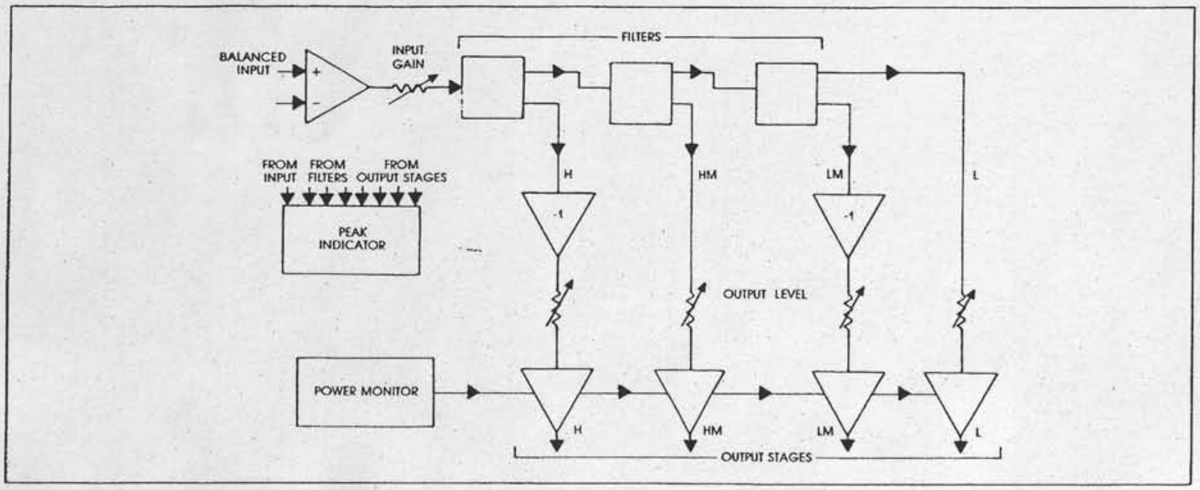


Figure 33 Block diagram of Ashly 4-way 12dB/octave crossovers

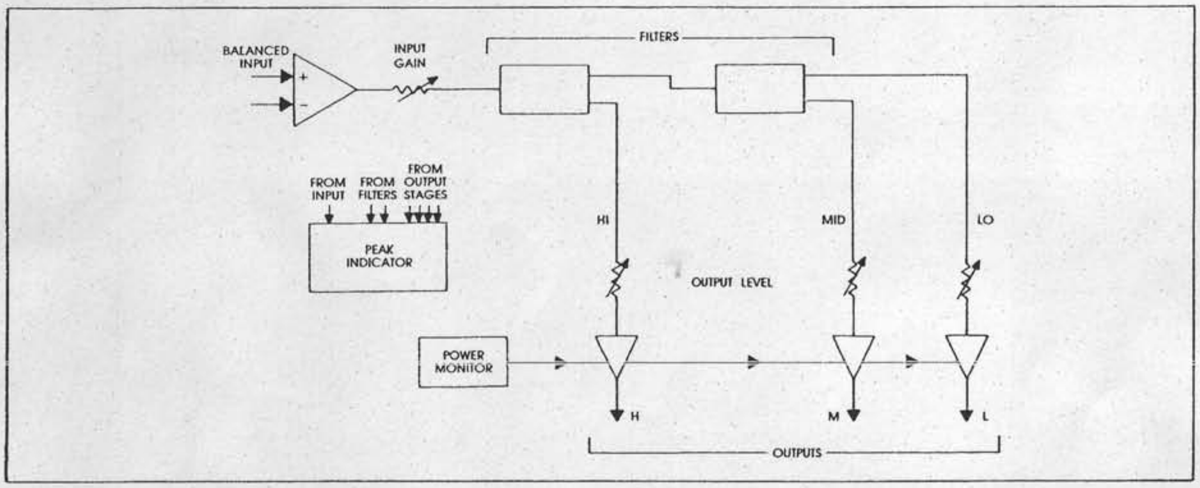


Figure 34 Block diagram of Ashly 3-way 18dB/octave crossovers

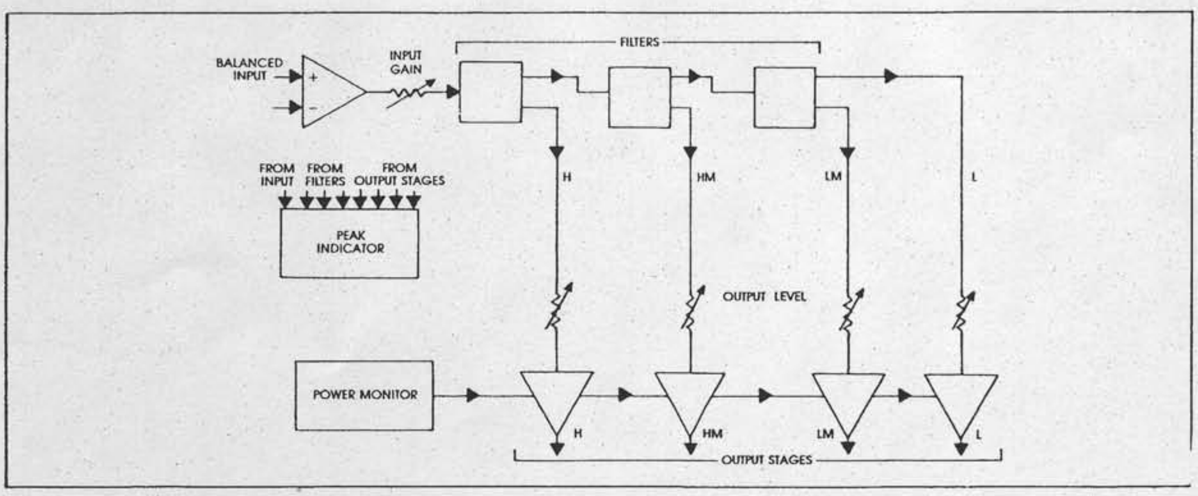


Figure 35 Block diagram of Ashly 4-way 18dB/octave crossovers