



**SC-50 / SC-55 LIMITER-COMPRESSORS
OPERATING INSTRUCTIONS**

ASHLY AUDIO INC. 100 FERNWOOD AVE. • ROCHESTER, N.Y. 14621
(716) 544-5191 • TOLL FREE (EXCEPT N.Y. STATE) (800) 828-6308

TABLE OF CONTENTS

	<u>PAGE</u>
1. INTRODUCTION/UNPACKING	3
2. FRONT AND BACK PANEL LAYOUT	4
3. INPUT, OUTPUT, AND POWER CONNECTIONS	5
4. EXPLANATION OF LIMITER/COMPRESSORS AND OUR CONTROLS	6
5. APPLICATIONS:	
SOUND REINFORCEMENT	16
RECORDING	20
BROADCASTING	21
SPECIAL EFFECTS	24
6. SPECIAL NOTES ON THE SC-55	26
7. BLOCK DIAGRAMS	27
8. CIRCUIT DESCRIPTION	27
9. DEFINITION OF TERMS AS USED IN THIS MANUAL	30
10. TROUBLE SHOOTING TIPS	33
11. SPECIFICATIONS	34

INTRODUCTION

The Ashly SC-50 Peak Limiter/Compressor was designed in response to the need for a universal peak sensitive automatic gain control (AGC) device with audio performance comparable to that of a professional mixing console. It took over five years of research to realize this goal, with much of that time spent in developing a wide-bandwidth, ultra-low distortion, low-noise VCA (voltage controlled amplifier). The resulting product is a versatile and highly listenable limiter/compressor suitable for use in professional sound reinforcement, recording and broadcasting.

The Ashly SC-55 Stereo Peak Limiter/Compressor was developed to meet the need for a stereo limiter/compressor that maintains correct stereo imaging at all times. Except for the addition of the BALANCE control and the absence of a DETECTOR PATCH, it is identical to the mono SC-50. The greater part of the text and illustrations in this manual therefore refers to the SC-50. Special features of the SC-55 are explained in the section, "Special Notes on the SC-55", on page 26. We ask that you please read this instruction manual thoroughly before operation so that you may realize all the features and benefits that the SC-50 and SC-55 limiter/compressors have to offer.

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.

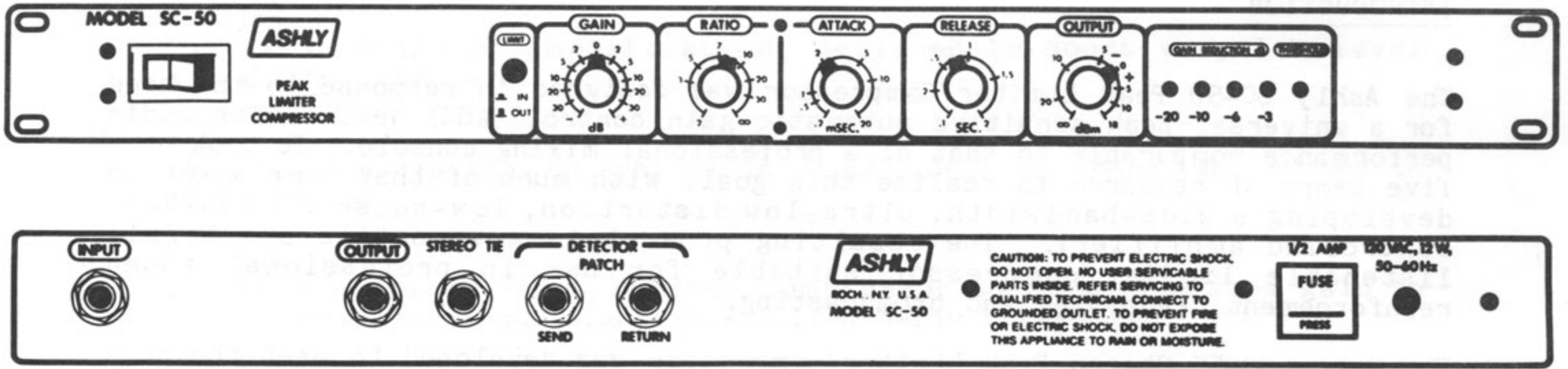


Figure 1 SC-50 front and back panels.

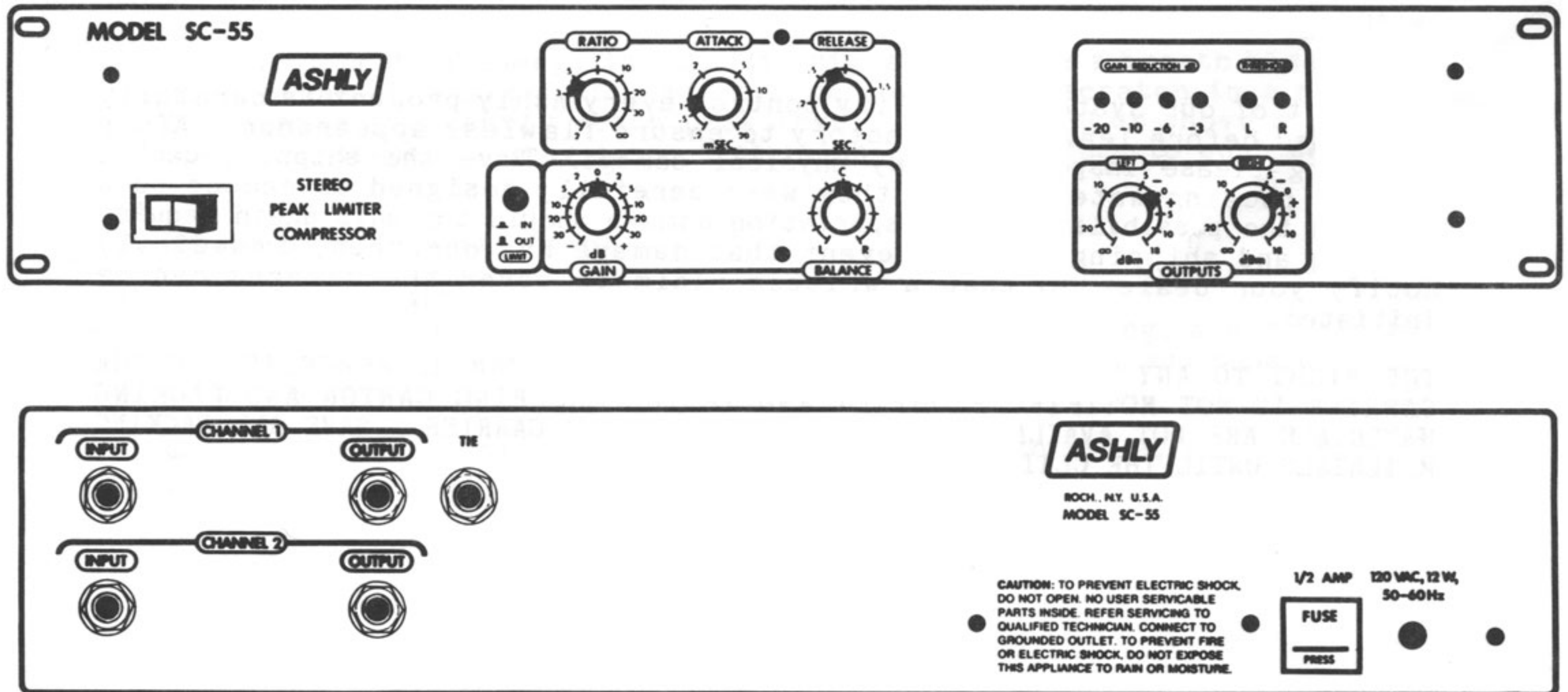


Figure 2 SC-55 front and back panels.

INPUT, OUTPUT, AND POWER CONNECTIONS

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

The INPUT (Fig. 3) is a 10k ohm active balanced type on a standard stereo phone plug that will accept levels of up to +20 dBm. The (+) or in-phase connection is on the tip and the (-) or out-of-phase connection is on the ring. When feeding the compressor/limiter 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 32.)



Figure 3 SC-50 Input Jack.

The OUTPUT connection (Fig. 4) is a standard 1/4" phone jack and mates with a standard phone plug such as a Switchcraft 280. Output impedance is 50 ohms, and full headroom is realized with any load of 600 ohms or greater. For rack mounted unbalanced audio systems the output ground may be separated from the case ground by using a stereo phone plug for the output connection. The output ground is then wired to the ring of the stereo plugs rather than the sleeve. In this manner, ground loops in the rack may be eliminated. This output can be fed to a balanced input by wiring the (+) input to the tip, the (-) input to the ring, and the shield to ground.

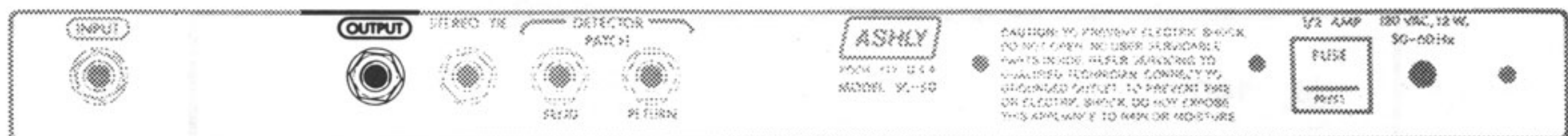


Figure 4 SC-50 Output Jack.

STEREO TIE

This patch point is used to tie together two or more SC-50's (or SC-55's) for stereo or multi-channel operation. To connect, place a patch cord between the STEREO TIE connections of both units. If signal ground isolation is being used, be sure to use a stereo patch cord.

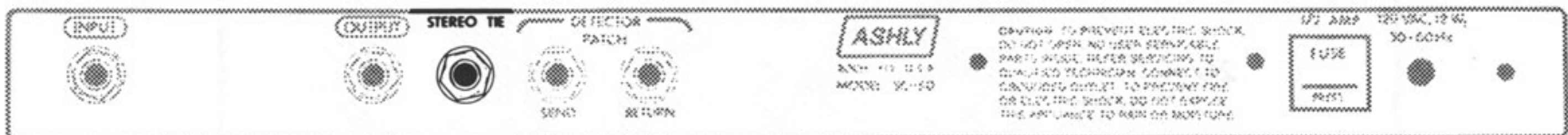


Figure 5 SC-50 Stereo Tie point, used to link two mono units for stereo operation.

NOTE: If more than 2 SC-50's are to be tied together, use a common "Y" type adaptor to split the signal as necessary.

The Stereo Tie point actually represents the DC output of the SC-50's detector circuit. When tied in parallel for stereo operation, the VCA's of two SC-50's will both respond to the most negative control voltage present at the Stereo Tie point, thus insuring that the stereo image never shifts left or right. The Stereo Tie points of SC-50's and SC-55's may be directly interconnected.

DETECTOR PATCH

The SC-50 Limiter/Compressor has a DETECTOR PATCH point (Figure 6) which can be used in conjunction with an equalizer to produce frequency sensitive limiting. This and other uses of the Detector Patch are discussed on pg. 20.



Figure 6 SC-50 Detector Patch Point.

EXPLANATION OF LIMITER/COMPRESSORS AND OUR CONTROLS

The Need For Gain Control

The human ear excels in its ability to detect an extremely wide range of loudness levels, from the quietest whisper to the roar of a Boeing Jumbo Jet! When we attempt to reproduce this dynamic range, by means of amplifiers, tape recorders, records, or radio transmitters, we run into one of the fundamental limitations of these electronic media: limited dynamic range. In some cases, such as amplifiers, the usable dynamic range is quite good, and is adequate for most musical program material. However, some audio equipment, such as cassette tape recorders, have a very narrow useful dynamic range (Fig. 7).

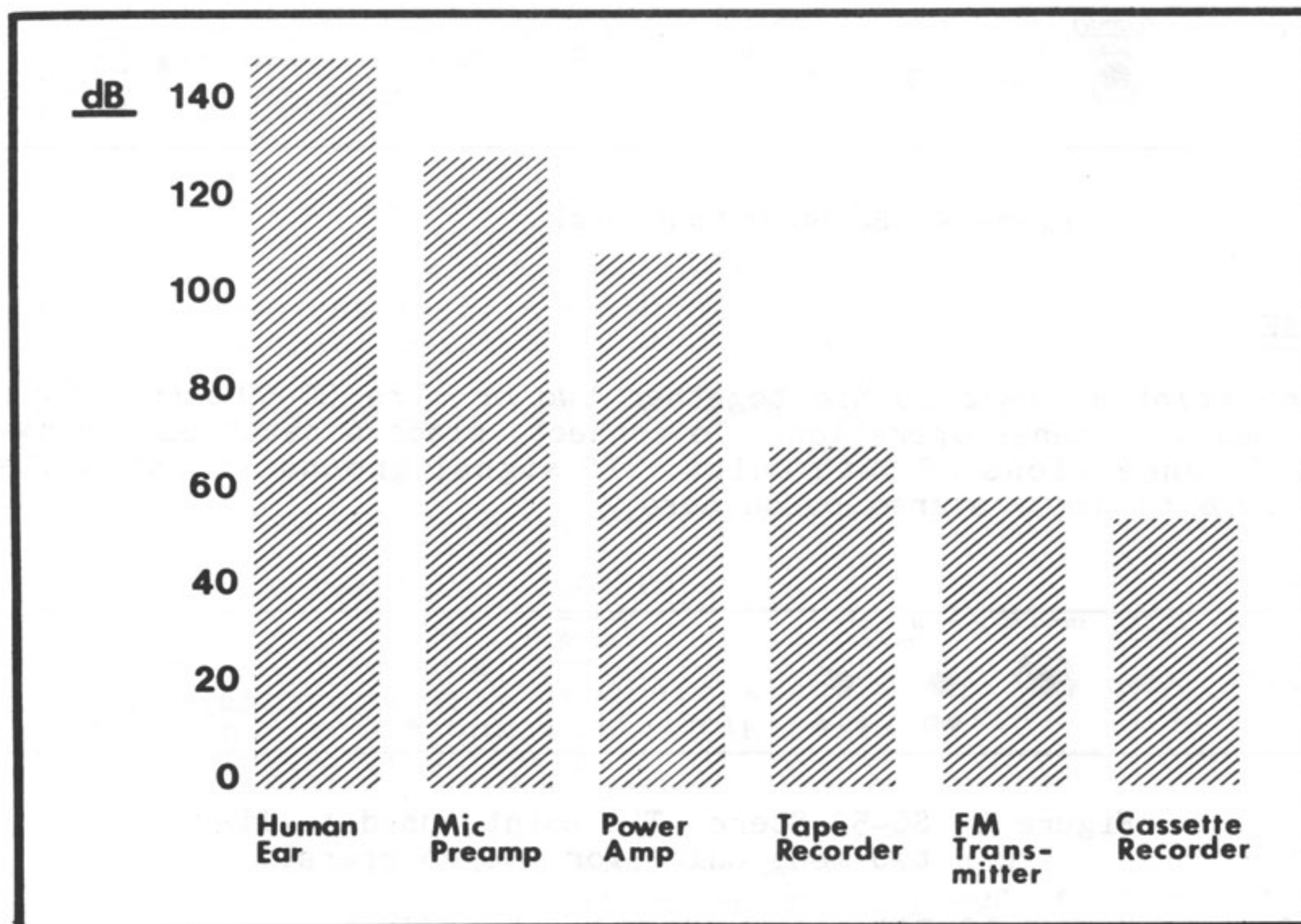


Figure 7 Comparison of the dynamic range of several common audio systems.

What is it that compromises the dynamic range of this equipment? As Fig. 8 shows, the useful operating region of a piece of audio equipment is squeezed in between noise and distortion. As program volume goes lower and lower, it approaches what is known as the "noise floor" of that particular piece of equipment, and if the volume of the program material goes lower still, it is engulfed by the noise. The noise floor, or minimum constant noise level of the equipment will consist of hiss, hum, transistor noise, record scratches, tape hiss, buzz and whatever noises are inherent in the medium. When the program level is considerably higher than the noise floor, our hearing system masks the noise, and it is not a problem. However, when listening to very quiet sections of a program, for example a pause between movements of a string quartet, the noise can become very bothersome.

At the other end of the loudness spectrum, the limitation on dynamic range is usually distortion, either in the form of amplifier overload or tape saturation. In most transistorized equipment, the transition from clean, undistorted operation to severe distortion is very abrupt. Therefore, it is common practice to operate a piece of equipment at a level that is somewhat below the distortion point, leaving a margin of safety for unexpected, transient volume peaks in the music. This safety margin is known as headroom, and may range from 10 to 25 dB. Lowering our standard operating level to leave ourselves some headroom helps prevent distortion, but at the same time it moves our average program level closer to the noise floor, thereby compromising signal to noise performance.

It becomes apparent that to get the most out of an audio system, you have to keep your standard operating level as high as possible without risking distortion.

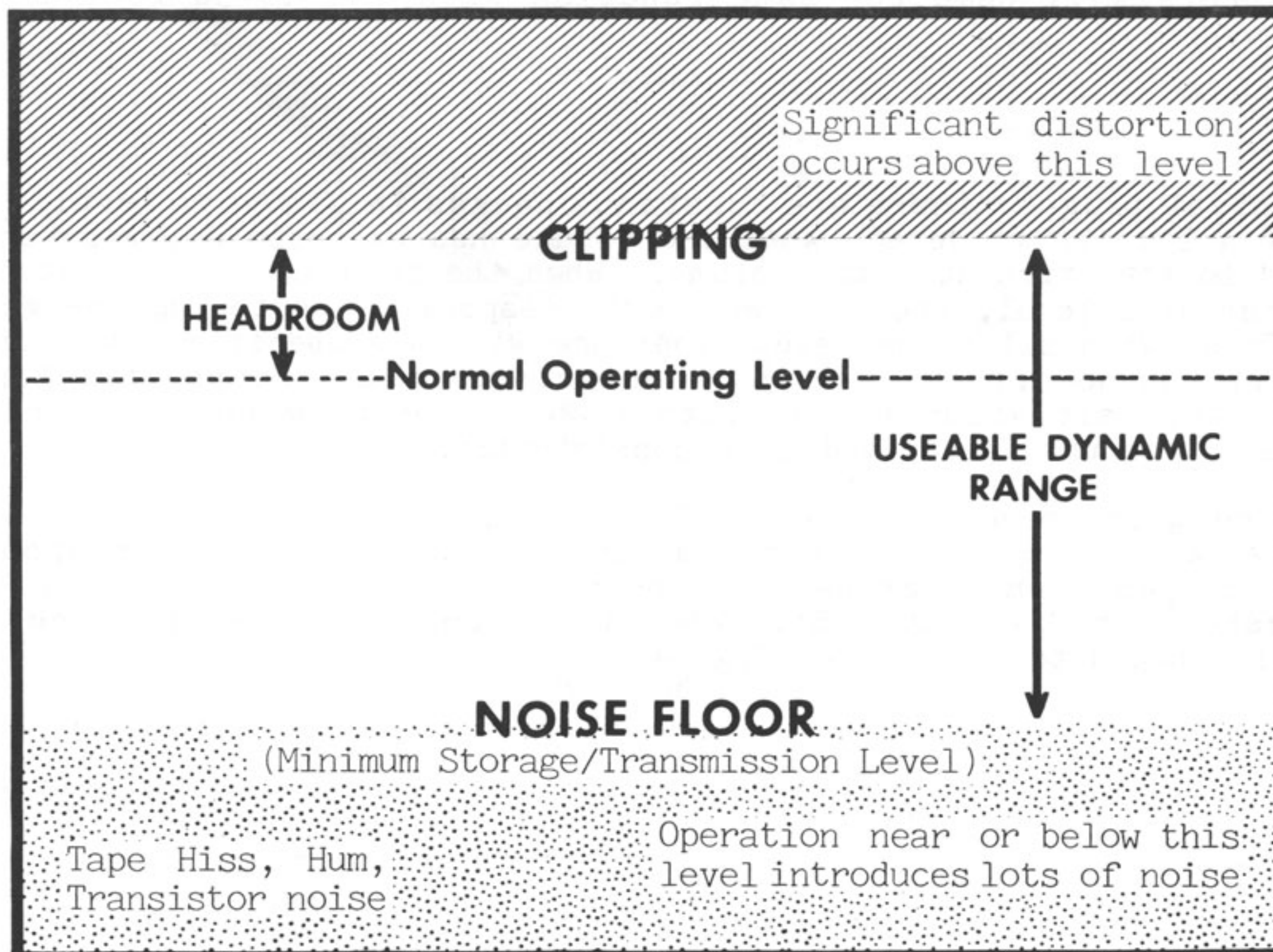


Figure 8 The useful dynamic range of any piece of audio equipment is limited by noise and distortion.

GAIN RIDING

One solution to the noise vs. distortion trade-off is to keep your hand on the volume control and manually adjust gain to suit the program. Indeed, there are times when this approach is entirely satisfactory. However, in most types of music there are instantaneous, short duration volume peaks, or transients, which would be difficult to anticipate and impossible to respond to with manual gain riding; you simply could not bring the volume down fast enough. In many situations, this can present real problems. For example, in recording, an extra burst of enthusiasm from a lead singer might overload the capabilities of your recording tape, causing ragged distortion and necessitating another take. If he felt it was his best performance ever, he won't be pleased to find out that it needs to be done over. In sound reinforcement, a sudden burst of energy through the system can blow fuses or even damage loudspeakers.

In addition to the problem of response time with manual gain riding, it also requires your constant attention, which takes you away from more important jobs. The need for a fast acting, reliable, automatic gain control is answered by limiters and compressors.

WHAT LIMITERS AND COMPRESSORS DO

Limiting and Compression are closely related effects, their differences sometimes being very subtle. Since limiting is generally an easier concept to deal with than compression, the following sections will talk about limiting first, followed by a discussion of compression on page 13. The use of the front panel controls is discussed in terms of limiting also. Bear in mind that these controls will have the same function when the SC-50 is used as a compressor.

LIMITING

In any musical program there are constant changes in loudness level. It is the job of a limiter to detect when the volume has exceeded a pre-determined level, and to then turn down the volume. When the incoming signal returns to its original safe level, the limiter should respond by restoring the gain to normal. Thus, when all volume excursions are within a specified "safe" range, the limiter has no effect. When an occasional peak occurs, the limiter responds. This situation is completely analagous to manual gain riding, except that it occurs faster and more consistently.

To determine which peaks are acceptable and which are too loud, the limiter establishes a reference level known as the threshold, and then responds to those volume peaks which exceed this point. In the Ashly SC-50 and SC-55, this threshold is fixed at 0dBV. When the input level reaches 0dBV, the limiter will begin to react. (see Fig. 9)

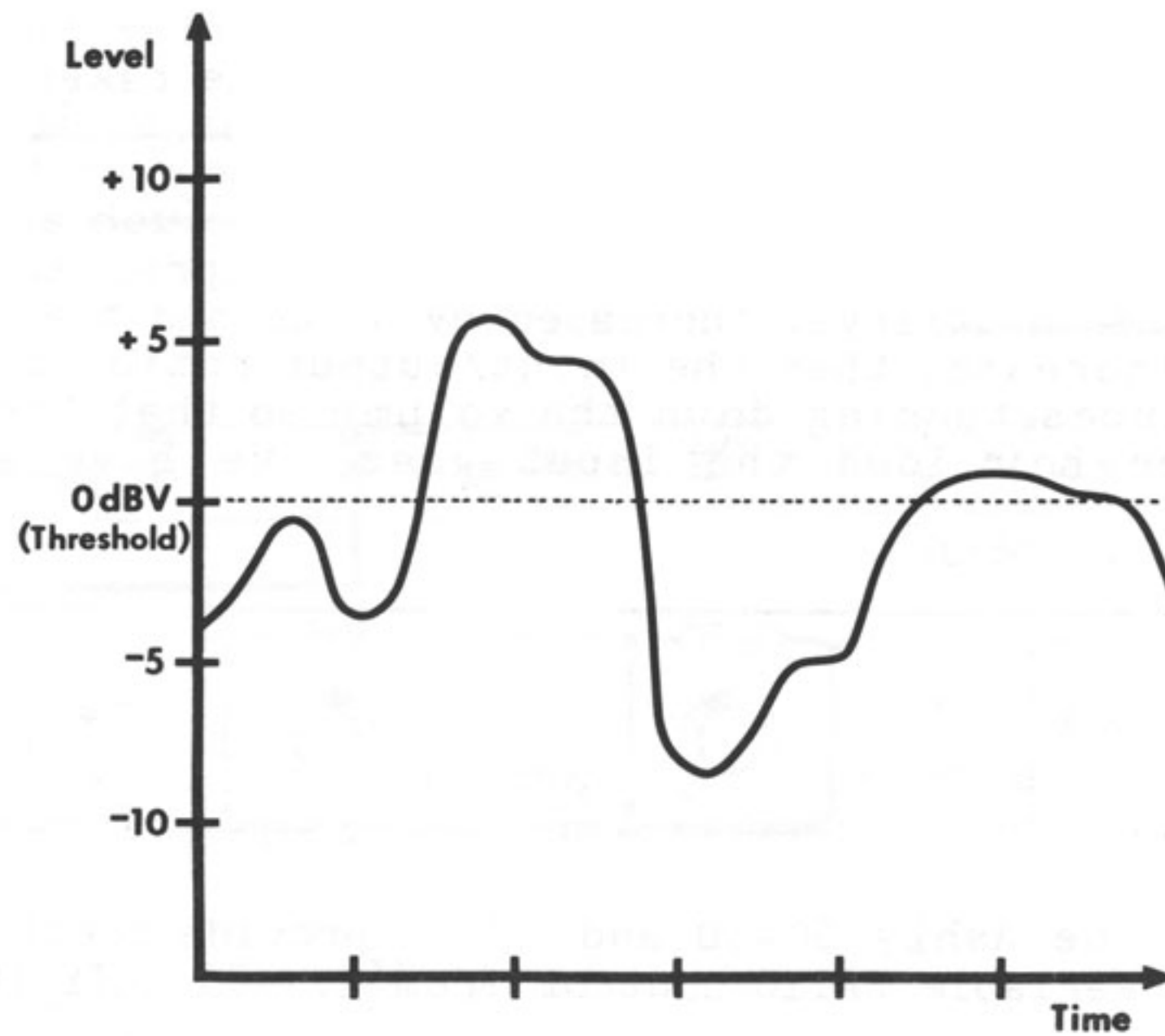


Figure 9 Changing loudness levels in a typical musical program. The louder peaks will be above threshold.

If the source feeding the limiter is increased in overall volume, then more of the volume peaks will now be above 0dBV; those peaks which were previously just below threshold level will now be at or above threshold, in addition to the volume peaks which had already been above threshold. Thus, varying the input level to the limiter is inversely proportional to changing the threshold level. On the Ashly SC-50 and SC-55, this effect is easily accomplished by changing the input GAIN control. Increasing the GAIN control is equivalent to lowering the threshold point. A gain control range of plus or minus 30 dB is provided.

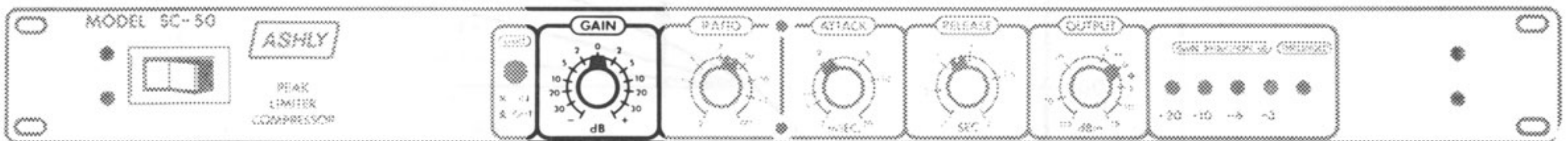


Figure 10 Varying the GAIN control changes the overall program level in relation to the SC-50's threshold.

What the limiter does once a peak above threshold is detected is determined by the RATIO, ATTACK TIME, and RELEASE TIME controls.

RATIO

This control determines the ratio of change in output level to changes in input level for all signals above threshold. The numbers printed around the ratio control are calibrated in dB and indicate the dB increase in input above threshold required to produce a 1 dB increase in output.

In the manual gain riding example, we stated that if an operator noticed that signal levels were approaching distortion, he would reach over and turn down the volume. The operator now has an option: he can turn it down so that it does not exceed his desired maximum level at all, or he can turn it down just

a little, so that the output level is slightly greater than his preferred maximum, but not as loud as it would have been had he taken no action. This can be expressed conveniently as a ratio. If he takes no action, then output volume will exactly follow input volume, a 1:1 ratio. If he turns down the gain so that the output level increases by half as much as the input level jumped, then input versus output changed by 2:1. Expressed in decibels, we would say that if the input level increased by 10 dB and the output change was limited to a 5 dB increase, then the input/output ratio is, again, 2:1. If our operator continues turning down the volume so that the output remains constant no matter how loud the input goes, we have an infinite (∞) input/output ratio.

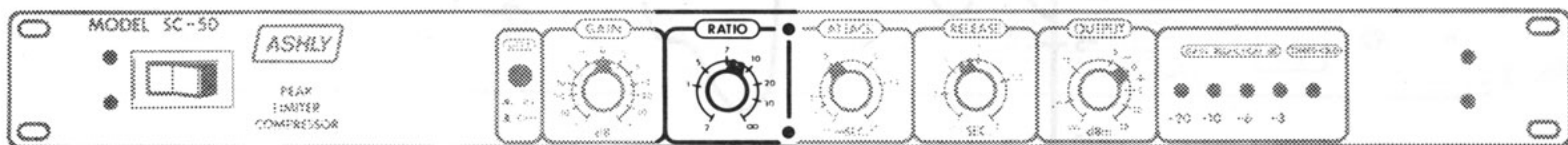


Figure 11 The Ashly SC-50 and SC-55 provide continuously variable RATIO control from 2:1 to infinity.

It should be remembered that the ratio control has no effect on signals which are below threshold, since the SC-50 is a linear amplifier at below-threshold levels.

The effect of the ratio control can be shown on a graph which plots input versus output (Fig. 12).

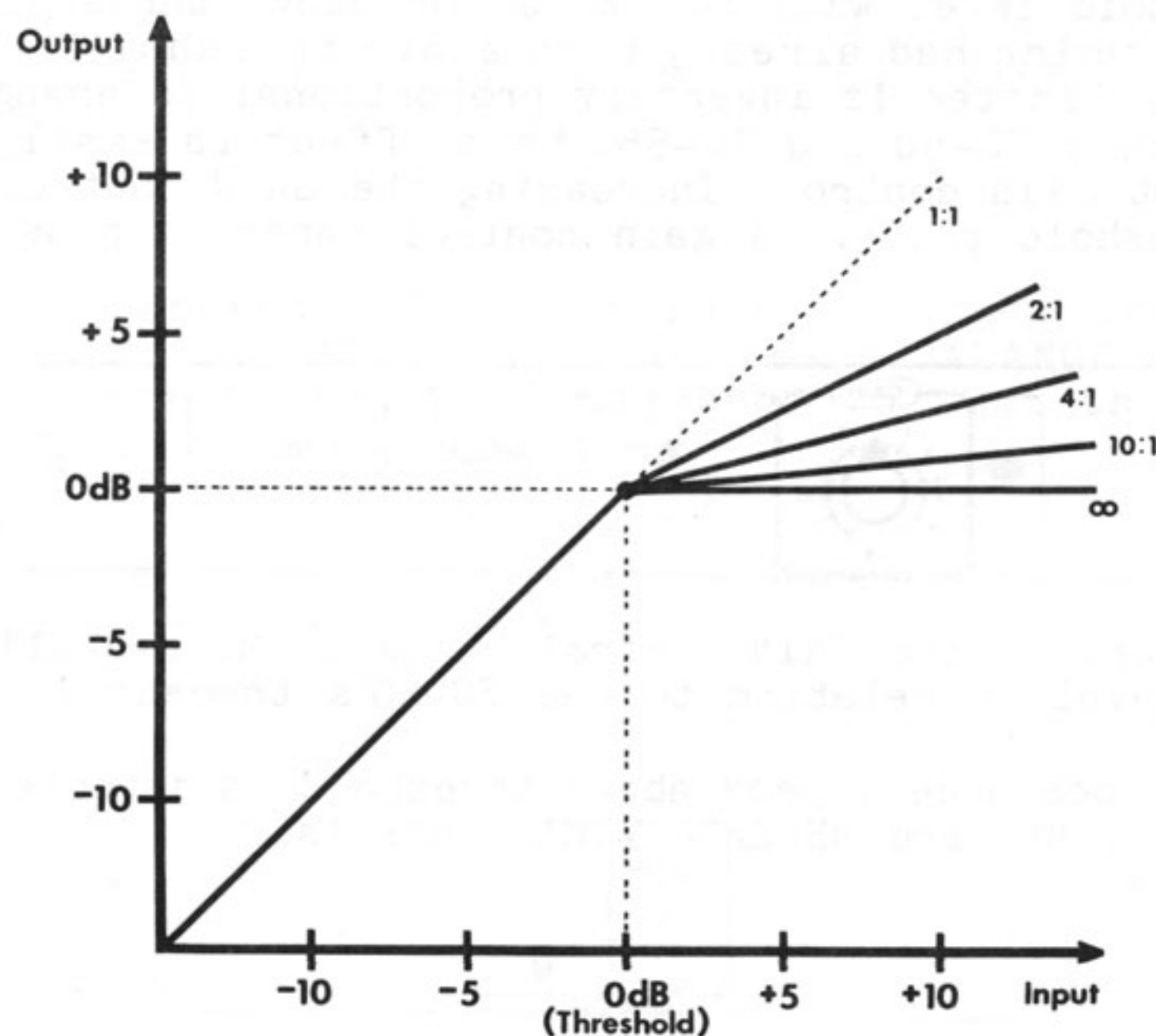


Figure 12 The effect of the RATIO control on output level. Larger ratios hold the output to lower levels.

There is a common but incorrect notion that limiting always implies the use of an infinite ratio. Although there are times when an infinite ratio is desirable, there will be situations where infinite, or "hard", limiting action is neither appropriate or necessary. In fact, it should be noted that an

infinite ratio setting is likely to cause noticeable side effects in the sound, and may not be usable on programs where subtle control is desired.

ATTACK TIME

The response of the limiter/compressor to signal levels above threshold is further defined by the ATTACK TIME control (Fig 13).

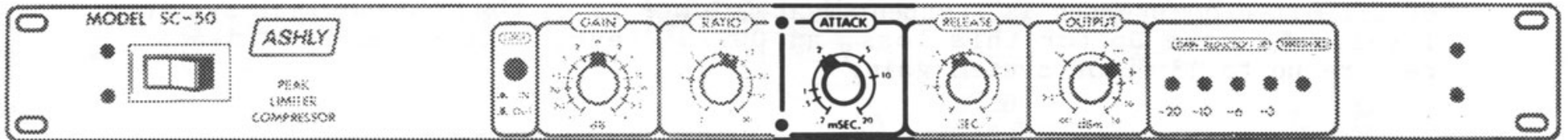


Figure 13 The ATTACK time control determines how quickly the SC-50 responds to signals which exceed threshold.

Attack time is the amount of time that elapses before the limiter begins to attenuate the output level after threshold has been reached. For very fast transients, such as hand claps, snare drums, or other percussive sounds, a very fast Attack time is usually desirable so that the limiter can respond in time to control the peak level. On other types of program material, a slower Attack time may be preferred. A too-fast attack may, on some material, "square off" the top of a waveform, producing a distorted sound.

The SC-50 and SC-55 provide continuously variable Attack times from 200 microseconds to 20 milliseconds.

RELEASE TIME

Another parameter which affects limiter performance is Release time, or the time required for the limiter to restore system gain to normal after the input signal has fallen below threshold level. Again, choosing Release time will depend on the type of program material being processed and the way in which the limiter is being used.

When subtle limiting is desired, slow Release times are often chosen to avoid a condition referred to as "pumping" or "breathing". This occurs when overall gain is being modulated up and down by repeated peaks which are followed by quieter intervals. If the Release time is set too fast, then the overall volume level will jump up and down, producing an objectionable and unsettling effect. Note that in some cases, an individual track or channel which seems to be pumping may sound acceptable when heard in context of a complete mix.

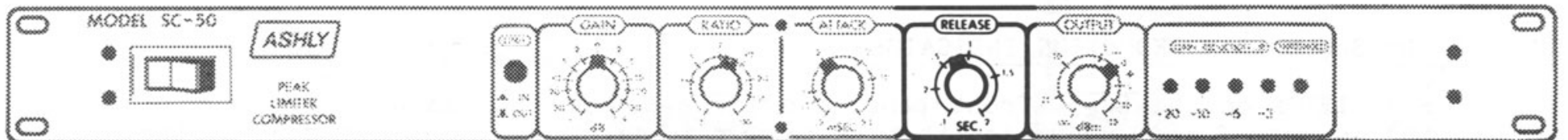


Figure 14 The RELEASE time control determines how quickly system gain returns to normal after a signal peak has passed.

A unique feature of the Ashly Limiter/Compressors is the incorporation of a double Release-time constant. When a conventional limiter is adjusted for slow release times, transients such as mic "pops" and record scratches may

cause a severe reduction in gain followed by a slow fade-up, making the action of the limiter very obvious. With the SC-50 and SC-55's double time constant, release from gain reduction after a brief transient is always fast, with a slower release after a sustained overdrive.

OUTPUT LEVEL CONTROL

Because limiting and compression are both gain reduction processes, the output of a limiter/compressor is frequently at a lower level than the nominal system level. To make up for this loss, an OUTPUT level control is provided to restore up to 18 dB of system gain.

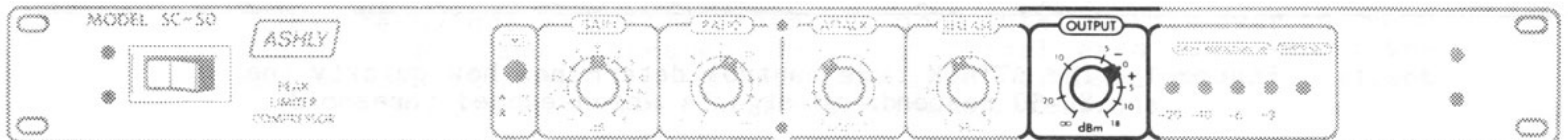


Figure 15 The OUTPUT control provides a convenient way of matching the SC-50's output to nominal system level.

NOTE: When the unit is in the BYPASS mode the OUTPUT control still functions.

BYPASS SWITCH

This switch enables you to quickly switch the SC-50 in or out of the audio chain, making comparisons between processed and unprocessed signals easy. When the switch is in the OUT position, all limiting and compression controls and functions are bypassed, with the exception of the output control, which continues to function as a straightforward volume control. For all normal limit and compress functions, this switch should be depressed to the IN position.



Figure 16 When the LIMIT switch is in the OUT position, all limiting and compression effects are by-passed. For normal operation, the switch should be IN.

THRESHOLD/GAIN REDUCTION INDICATORS

Five LED's on the SC-50 front panel provide a convenient visual indication of the amount of gain reduction that is taking place at any time. As soon as the OdBV threshold level is reached, the yellow LED illuminates. Depending on how far the input level rises above threshold, and the settings of the RATIO, ATTACK, and RELEASE controls, successive red LED's will illuminate, indicating increasing gain reduction.

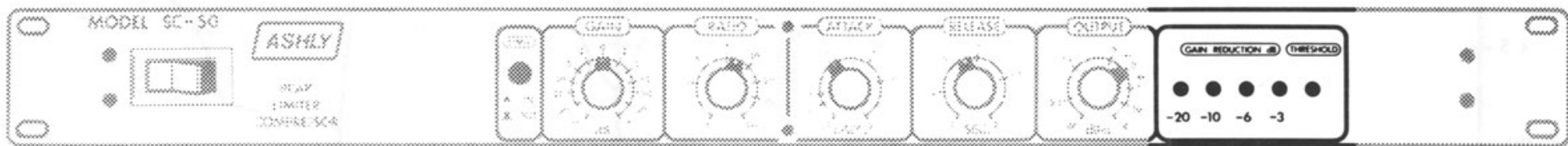


Figure 17 An LED display indicates when THRESHOLD has been reached, and also how much GAIN REDUCTION is taking place.

Gain reduction is a useful way of expressing limiter/compressor action. We have seen that the output level of a limiter is less than the input level by some amount. Using an infinite limit ratio, output level will remain nearly constant as input levels increase above threshold. With a gentle ratio, say 2:1, input signals above threshold will be "gain-reduced" at the output by exactly 1/2. Thus, gain reduction can be expressed as input level minus output level. For example, a +12 dBV input signal that is 3:1 limited will produce 4 dB of change at the output, and so we say that 8 dB of gain reduction has occurred (12 dB input minus 4 dB output = 8 dB gain reduction). The -6 dB red LED on the SC-50 front panel would be seen to light up.

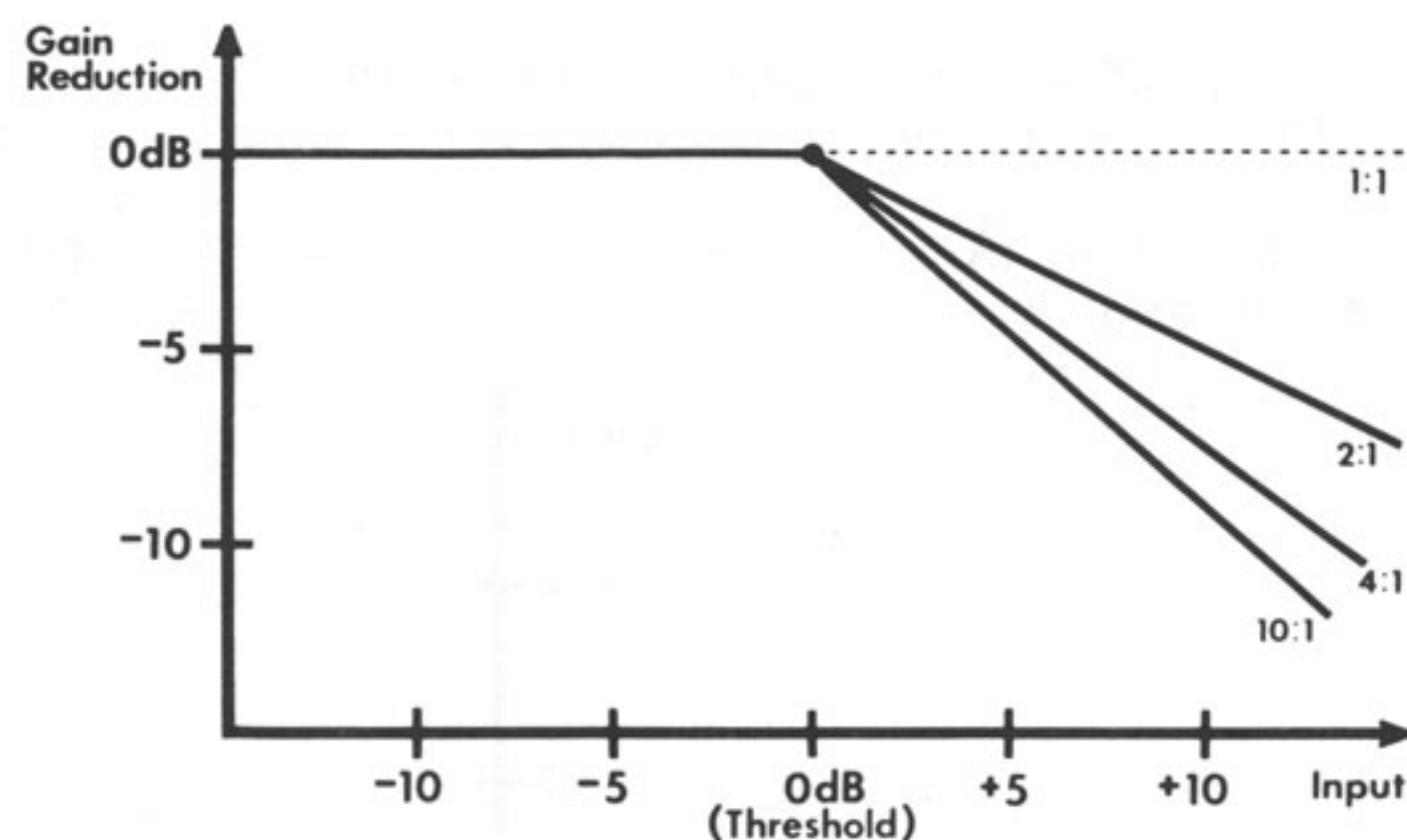
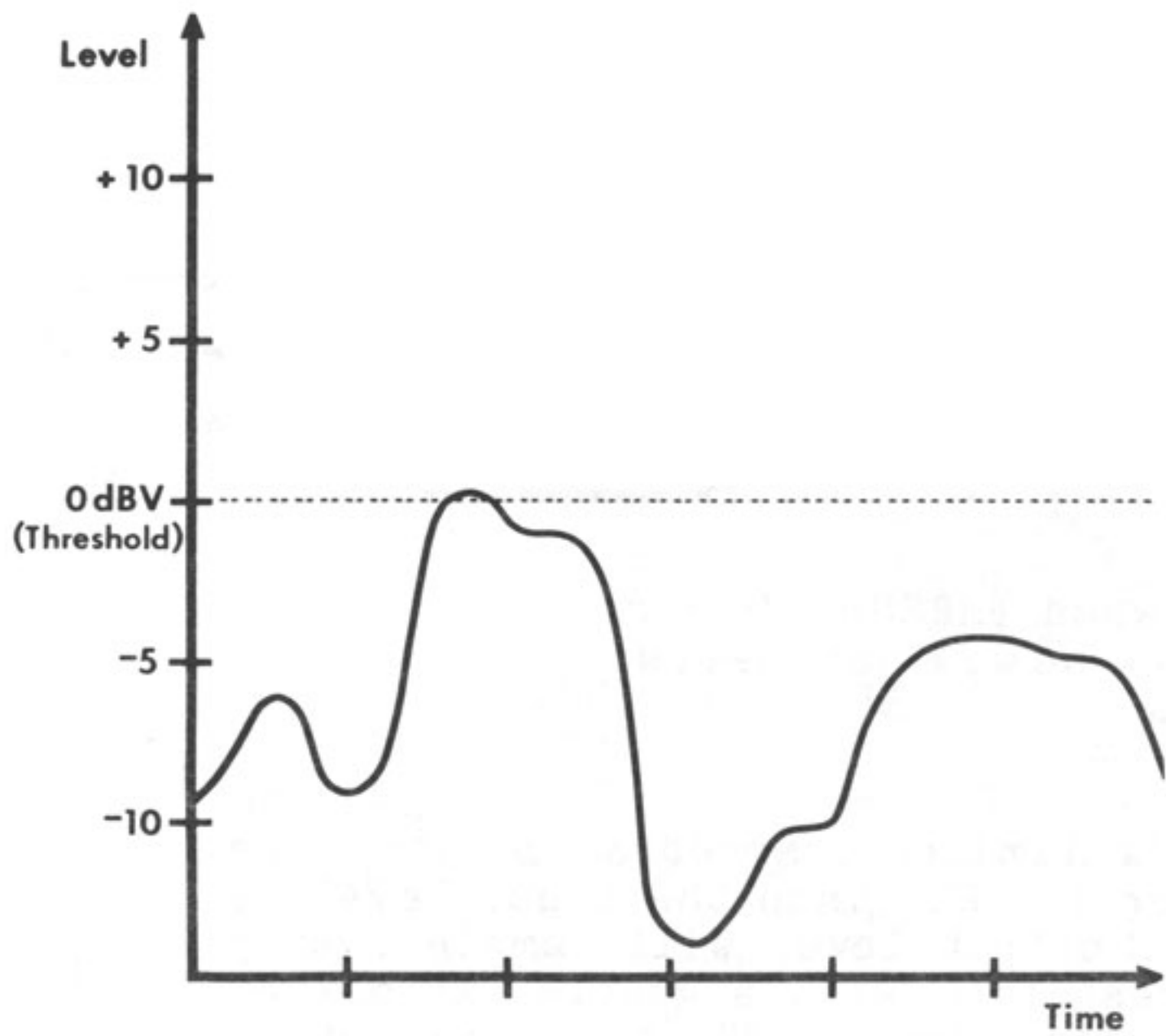


Figure 18 The effect of a limiter/compressor can be expressed as the amount of GAIN REDUCTION that is taking place for any given INPUT level.

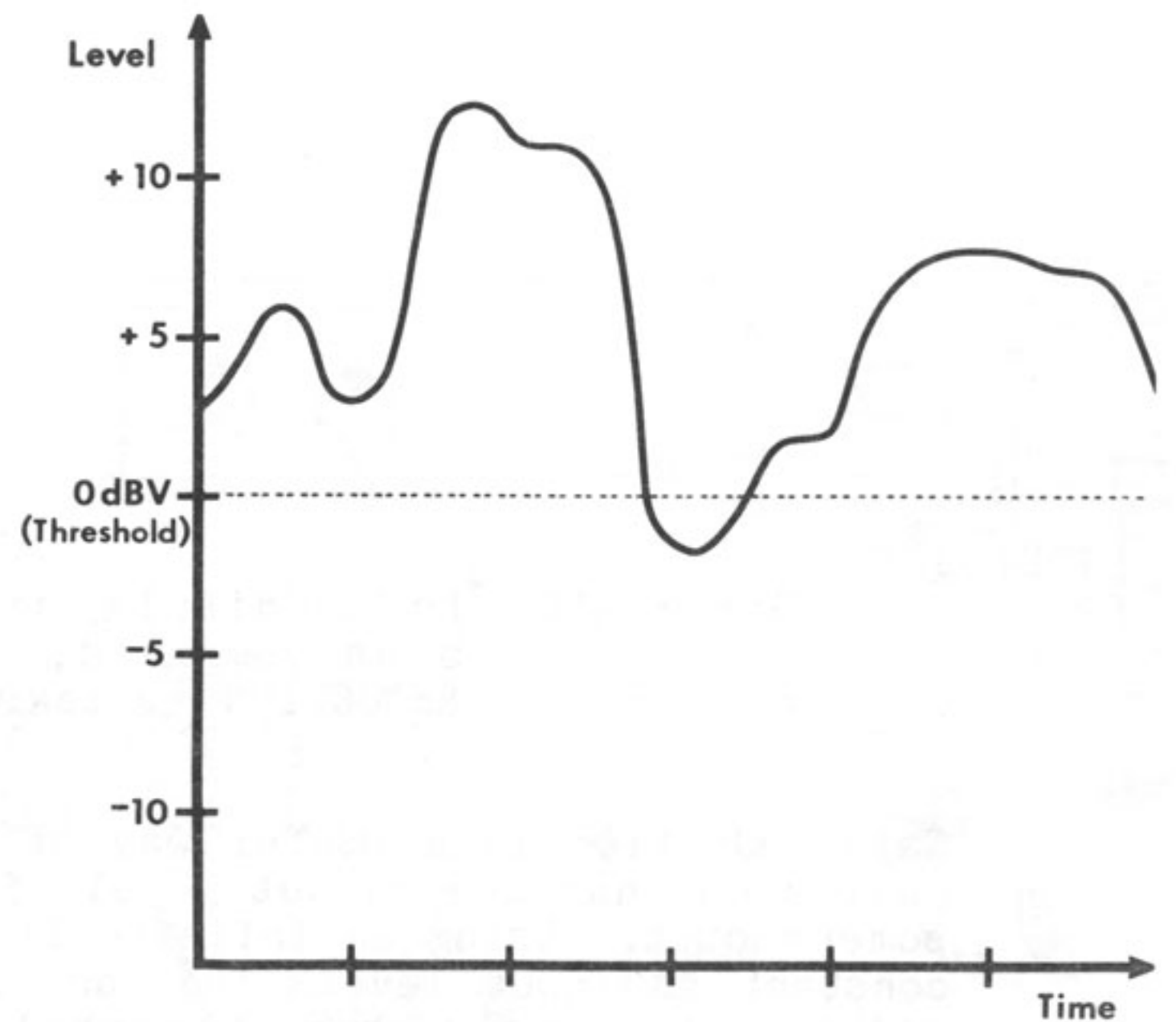
COMPRESSION

There is a good deal of disagreement and confusion in the audio industry over the definitions of limiting and compression, and the terms are frequently and incorrectly swapped back and forth. The fact is, compression and limiting are closely related effects, and the differences between them are largely a matter of degree. Indeed, it could be said that the difference between compression and limiting is more often than not a matter of what you intend the device to accomplish than the way it actually ends up working. This is because of the widely varying nature of musical programs.

Returning to our original definition of limiting, we stated that gain reduction occurred when input signals rose above a pre-determined threshold level of 0 dBV, and that the effective threshold could be varied by changing the GAIN control. Figure 19 illustrates two identical input signals, with one having the GAIN control increased by 12 dB.



(a)

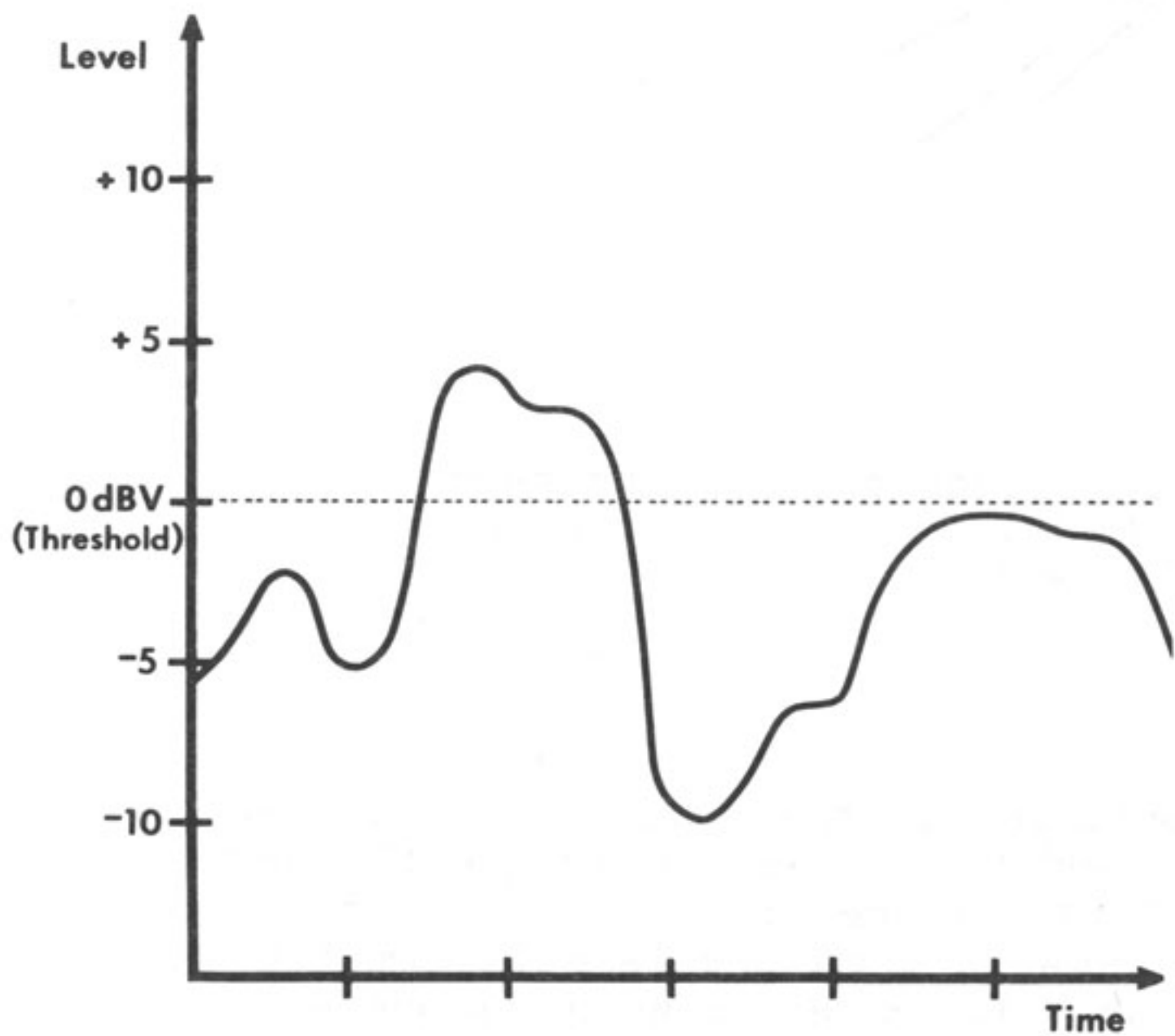


(b)

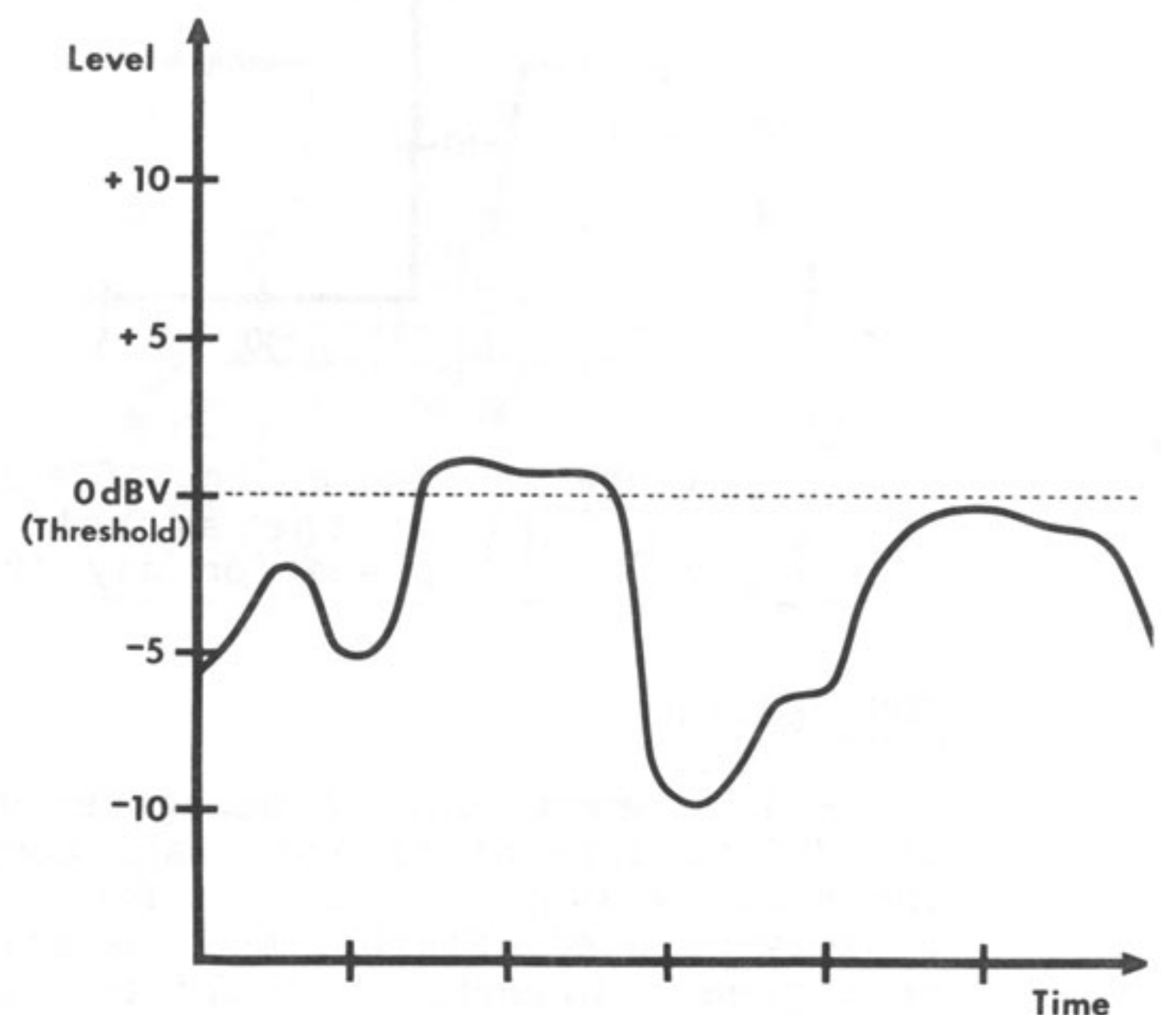
Figure 19 The constantly changing loudness levels of a typical musical program. The over-all volume level of the signal on the right is greater; it is above threshold most of the time. This is in contrast to the graph on the left (a), where the loudest volume peak just barely reaches threshold.

We know that a limiter only affects those signals which are above threshold, so the signal in Figure 19 (b) will be affected more than in Figure 19 (a).

Assuming an identical ratio for both cases, look at what happens to the two input signals in Figures 20 and 21.

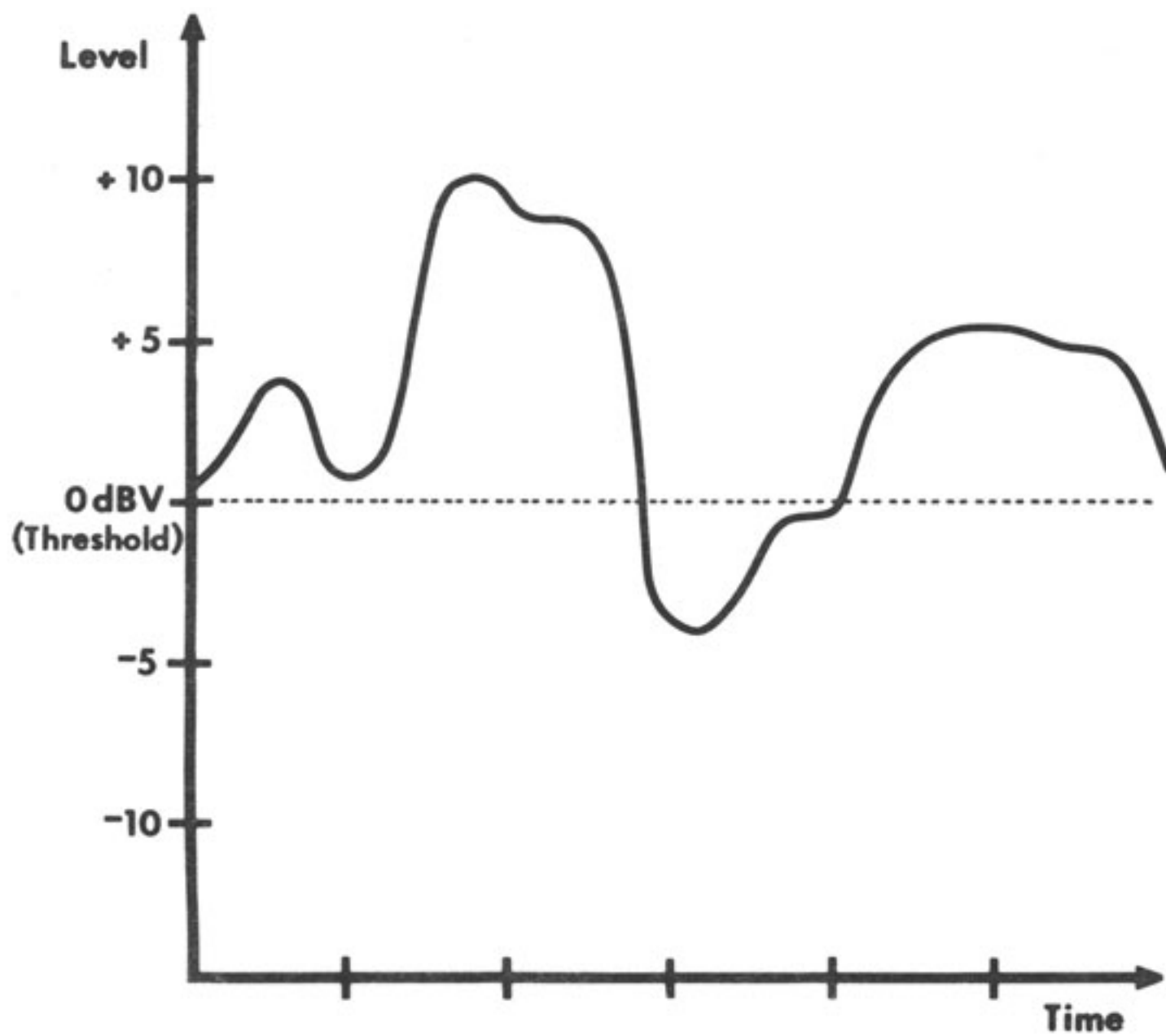


(Input)

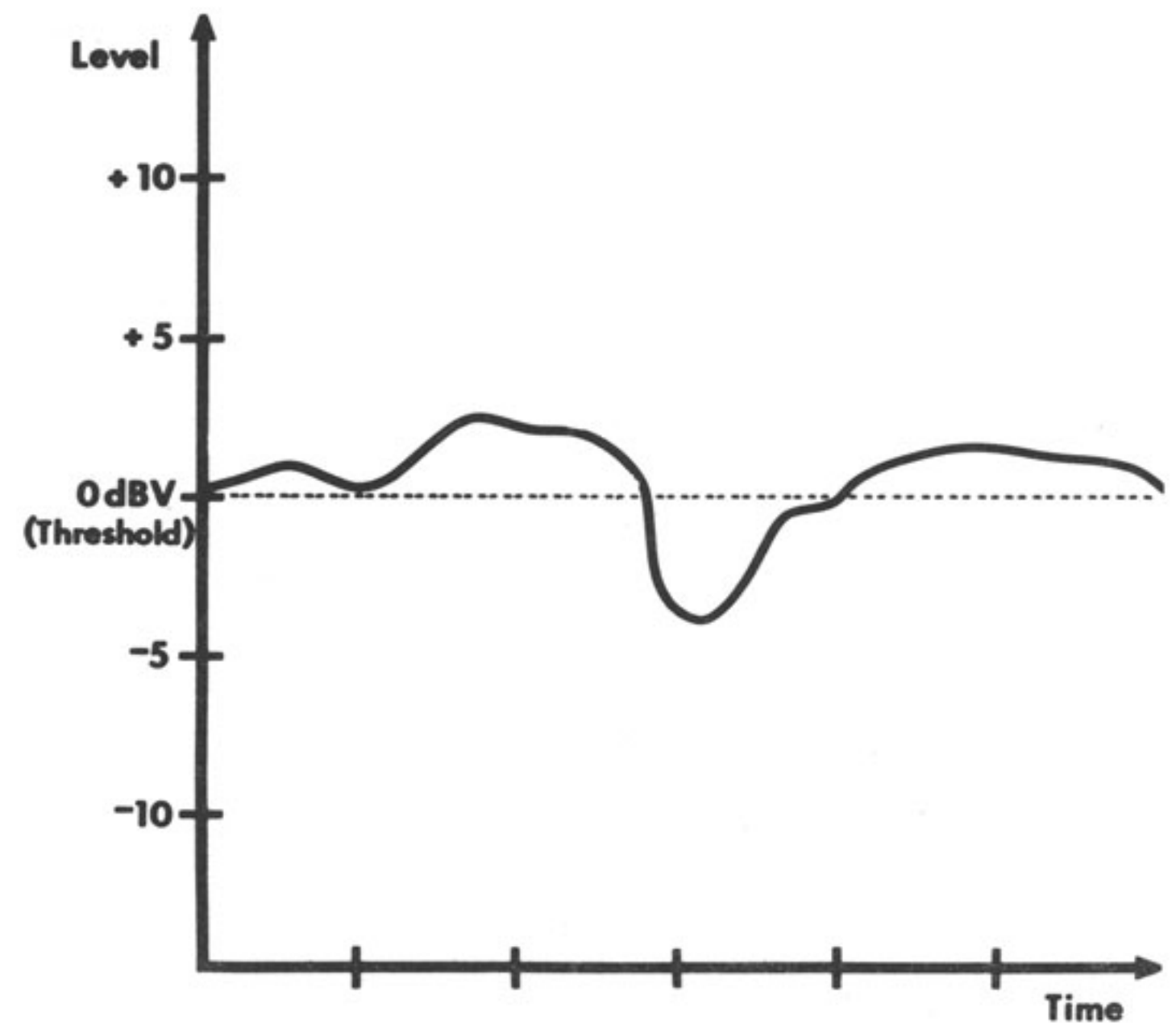


(Output)

Figure 20 LIMITING. Once the Input level begins to exceed threshold, limiting will begin to occur. Notice that the Output is an exact copy of the Input for all signals which are below threshold. Only the signal peak which exceeded threshold has been gain reduced, in this case by a 4:1 ratio.



(Input)



(Output)

Figure 21 COMPRESSION- The overall input level has been increased by changing the GAIN control, so that now the overall signal level is above Threshold most of the time. Notice that the graph of the Output signal has undergone a substantial change from its input appearance. The compression ratio is still 4:1, just as in Fig. 20, but considerably more gain reduction has occurred.

As Figures 20 and 21 show, a very significant difference in dynamic range is achieved simply by changing the relationship between nominal signal level and threshold, as a result of varying the GAIN control. The most interesting effect to be noted, however, is seen by comparing the original Input signal of Figure 19 (a) with the Output signal of Figure 21. Notice that the quietest portions of the original signal have effectively been increased in volume while the loudest portions of the original signal have been decreased in volume. In effect, both ends of the dynamic spectrum have been pushed toward the "middle". This is quite different from simple limiting, where only loud peaks are subjected to gain reduction. More than anything else, it is this double-ended effect which distinguishes compression from limiting.

Remember, in all of these examples, the only control which was changed was the GAIN control. The RATIO, ATTACK TIME, and RELEASE TIME were assumed to be identical in all cases. The only factor which turned limiting into compression was a change in the relationship between input level and threshold.

Compression is further differentiated from limiting by careful selection of Attack and Release times.

When limiting is employed to protect an audio system against transient volume peaks and possible overload, Attack time is usually set as fast as possible, consistent with distortion-free performance. Release time would also be relatively short, so that the output signal would be restored to normal as quickly as possible after the transient.

Compression is frequently used to keep overall signal level within a specific

dynamic range, and for this application, slow Attack and Release times are usually chosen. This approach is analagous to our manual gain riding example, where our operator is fading the music up and down to keep it fairly constant, but is doing it slowly enough so that the listener is unaware that the gain is being altered.

It should now be clear that the Ashly SC-50 and SC-55 can function as either a limiter or a compressor, due to the wide range of control parameters which are available to the user.

APPLICATIONS

The applications of the Ashly SC-50 and SC-55 Peak Limiter/Compressors can be divided into two basic categories; it may be used as a protective device to prevent audio levels from overloading associated systems, such as tape recorders, amplifiers, speakers, or transmitters, or it may be used to create special effects and unusual sounds for recording and musical performance. These two different approaches to using the limiter/compressor impose vastly different and contradictory demands on the unit's performance.

When used in a protective mode, the unit is usually required to control the dynamic range of an incoming signal, and to do so without audible side effects. Correctly adjusted, the listener should be unaware of the limiter's presence.

In the early 1960's, when musicians began looking at the recording process as a way to create new sounds, the pumping effect which had been avoided like the plague by earlier engineers was suddenly seized upon and utilized as a creative tool, laying the groundwork for many of the sounds which are now considered indispensable in contemporary music. In this role, the compressor is used because you can hear it working, and control of dynamic range is only a secondary consideration.

The Ashly SC-50 and SC-55, with their wide range of control parameters, are well suited to both of these applications.

THE LIMITER/COMPRESSOR AS A PROTECTIVE DEVICE

Sound Reinforcement

The SC-50 and SC-55 provide fast and accurate gain control for the prevention of sound system overload due to unexpected transients. Sound system distortion is usually a result of amplifiers running out of power, in which case nice round waveforms turn into harsh-sounding squared off waveforms. Looking at it from the perspective of a speaker diaphragm, this means that, whereas in normal operation the diaphragm is required to accelerate, slow down, smoothly change direction, and accelerate again, distorted operation requires an instant acceleration, instant stop, a change of direction, and instant acceleration again. Since speaker diaphragms are subject to the laws of physics, they won't take this kind of punishment for long. The diaphragm may shatter, or its voice coil may overheat. In addition to the damage caused by sustained overload, the speaker may also be damaged by occasional, one-shot high level overload, for example, the sound of a microphone falling face-first into a hard floor. Even if this type of transient doesn't destroy a speaker outright, it may damage the speaker surround in such a way as to cause mechanical abrasion and future failure.

INSTALLATION

To install the SC-50 in a sound system using a passive crossover, insert it between your mixing console output and the power amplifier input. (Fig. 22)

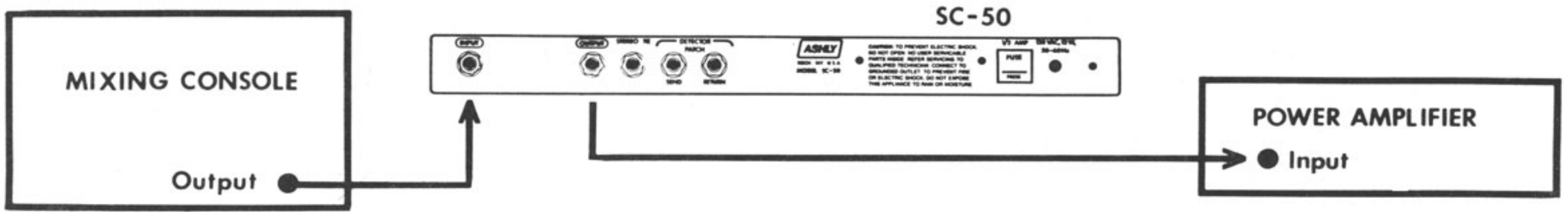


Figure 22 Installing the SC-50 for use in a sound system with a passive crossover.

For systems using electronic crossovers, there are two ways to use the SC-50. As shown in Fig. 23, the SC-50 may be inserted between the console output and the crossover input, in which case it will act on the entire audio frequency spectrum.

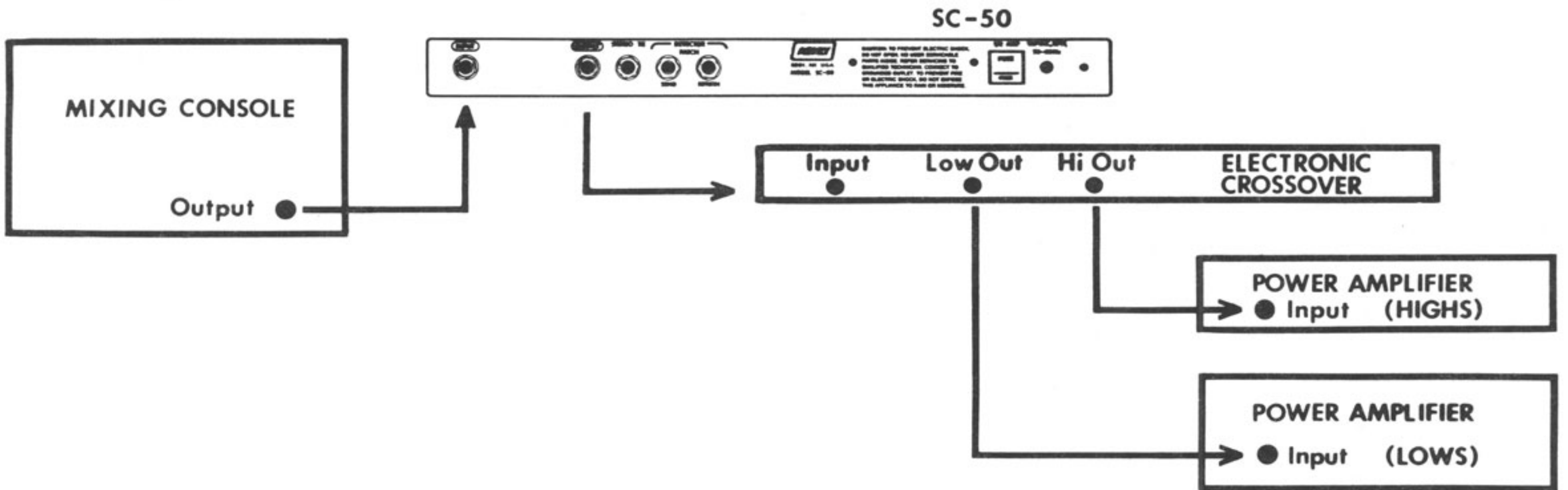


Figure 23 The SC-50 in a biamplified system.

Alternately, the SC-50 may be inserted between an output of the electronic crossover and the input of a power amp, in which case it will only affect a specific band of frequencies. This is shown in Fig. 24.

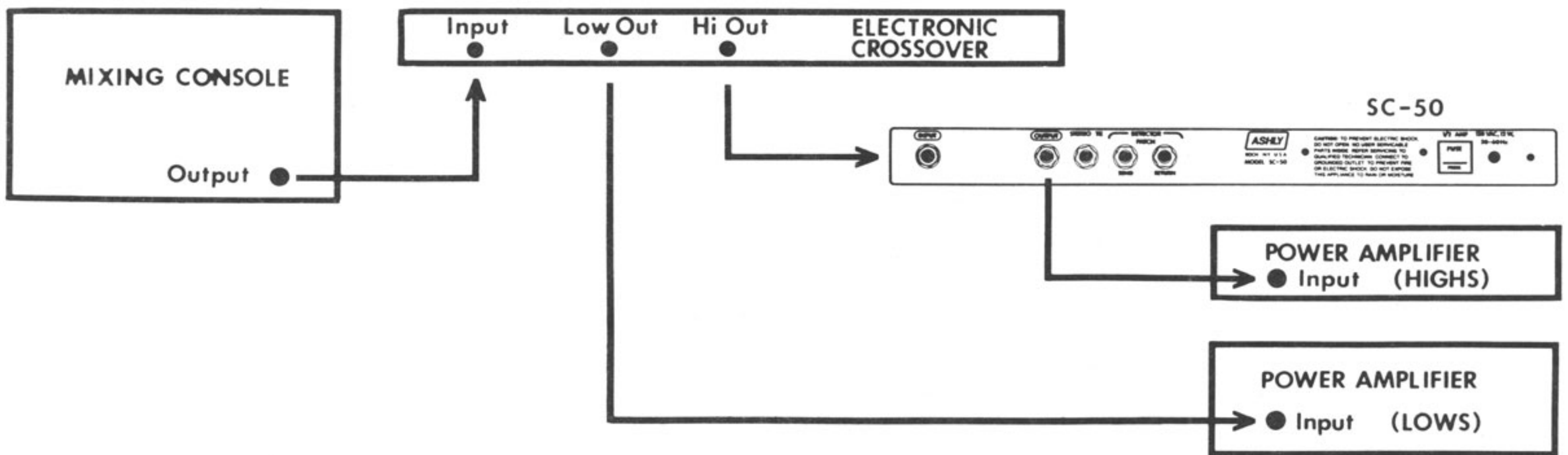


Figure 24 Alternate biamplification approach.

SET-UP PROCEDURE

Exact control of system headroom requires defining the clipping point of your amplifiers in relation to the nominal zero VU of your mixing console. A signal generator and oscilloscope would be helpful in determining this point. Since many people do not have the facilities or time to set up their systems in this way, an alternate approach is suggested.

First, realize that your sound system may have several different "0 dB" levels, especially if equipment of different manufacturers is being mixed. 0VU on your mixer's meters may or may not correspond to a 0 dBV level of .77 Volts. Input sensitivities of power amplifiers also vary. However, you don't have to carry around a notebook full of specifications to use the SC-50 in your system.

As an example of a typical set-up procedure, refer again to the signal chain shown in Fig. 23.

1. Begin by turning the crossover level controls down.
2. Switch the SC-50 LIMIT switch to the OUT position.
3. Set both the GAIN and OUTPUT level controls to 0 dB.
4. Set the RATIO control to 10 and adjust the ATTACK and RELEASE controls to a relatively fast setting.
5. Plug a good quality music source into your mixing console and adjust the console so that its meter peaks at around 0 VU.
6. Look at the LED display on the SC-50 and adjust the GAIN control so that the yellow THRESHOLD LED just lights on peaks.
7. Note the new position of the GAIN control and adjust the OUTPUT control inversely. That is, if you increased the Gain control to +10 dB in order to light the Threshold LED, then decrease the Output control to -10 dB. Or, if you had to decrease the Gain control to just light the Threshold LED, then increase the Output by an equal amount. You have now gotten your console and the limiter/compressor into relative agreement over what is a maximum permissible signal level; as long as your console's output meter stays out of the red, little or no limiting will occur. At or above console 0 VU, limiting will take place.
8. Turn the input volume control of your electronic crossover up to its nominal zero point (for Ashly crossovers this is number 7). Turn the OUTPUT level controls of the crossover up until you achieve a good musical balance of low and high frequencies, and continue to increase the output volume controls until your sound system just begins to sound a little distorted. Since audible distortion may already be around 5%, it's a good idea to back off the output volume controls just a touch after you first begin to hear any distortion.

All elements of your sound system are now in agreement. Console 0VU = Limiter Threshold = onset of clipping in your power amplifiers. If you wish to allow extra headroom in the system between console 0VU and the onset of clipping, turn down the SC-50's OUTPUT volume control by 5 dB, 10 dB, or whatever safety margin you desire. (You can turn the music down now.)

9. Start with a 1 ms. ATTACK time, a .5 sec. RELEASE time, and a RATIO of 10. Adjust these parameters according to the type of program material, and firmness of control desired.

10. Activate the limiter/compressor by depressing the LIMIT IN/OUT switch. Now, loud bursts of energy above 0 VU will cause little or no clipping in your system.

COMPRESSION AS AN AID TO NOTCHING OUT FEEDBACK

A common ritual in sound system set-up is equalizing the room to remove feedback. This is generally accomplished by turning up system gain to purposely induce feedback, searching for the center frequency of the feedback, and then equalizing at that frequency to remove the feedback. Once this frequency has been cut, system gain is again increased to induce another feedback point, and the whole procedure is repeated until the engineer is satisfied that the significant problem frequencies have been corrected. The major problem with this approach is that the feedback can easily get out of control, and the engineer ends up dashing back and forth between the mixer volume controls and the equalizer controls, while everyone in the room plugs their ears and prays it will end soon. The Ashly SC-50 can turn this procedure into a fast, painless job, eliminating loud feedback levels and the possibility of speaker or ear damage.

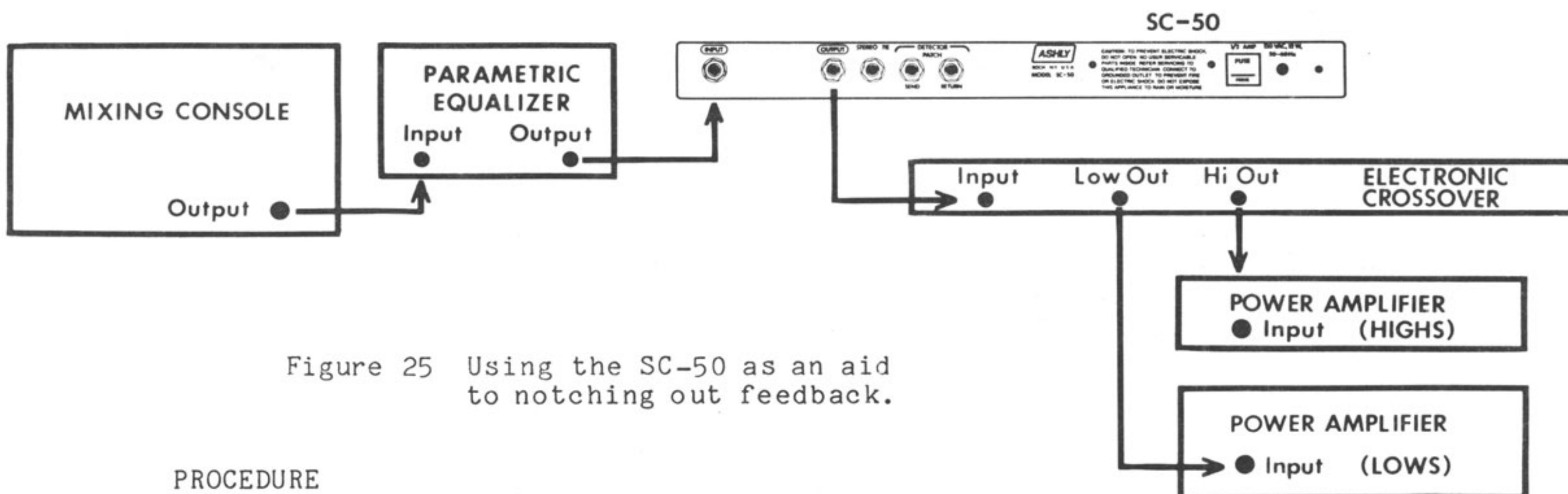


Figure 25 Using the SC-50 as an aid to notching out feedback.

PROCEDURE

1. Set up the SC-50 controls as follows:
 - a. OUTPUT level control to -20 dB.
 - b. Input GAIN control to maximum.
 - c. RATIO control to infinity (∞).
 - d. ATTACK time to 5 ms.
 - e. RELEASE time to 1 sec.
 - f. LIMIT IN/OUT switch IN.
2. Adjust equalizer controls to a flat setting, and if the equalizer has an overall volume control, boost it by 10 to 15 dB.
3. Open up several microphone input channels to a normal operating level, with typical EQ settings, and turn the console master fader up to a louder than normal setting.

At this point, the system should be well into feedback, but the room volume will remain constant due to the action of the compressor. You can listen to the feedback at any level you like by simply varying the SC-50's OUTPUT volume control, although below a certain monitoring level, the feedback will stop.

4. Try to determine the feedback frequency, and then equalize it by adjustment of the center frequency, bandwidth, and boost/cut controls of your parametric equalizer. (Note: a graphic equalizer can also be used, with a decrease in accuracy.) After eliminating the problem frequency, try to further define it by sharpening up the bandwidth, re-tweaking the frequency control, and making the cut shallower, if possible.
5. As soon as the first feedback frequency has been removed, the Ashly Sc-50 will automatically bring up system gain until another feedback point is induced. Repeat the equalization procedure until it becomes impossible to distinguish individual, predominant feedback frequencies.
6. Return all mixer, EQ overall gain, and limiter/compressor gain controls to normal settings.

RECORDING

The Ashly SC-50 can be used to prevent saturation of magnetic recording tape and to control tape hiss. In professional recording studios, the saturation level of the tape, system headroom, and the output level of the console are all known quantities, making the application of limiting and compression very easy.

An example of the use of limiting to prevent tape saturation and improve signal to noise performance was given in the earlier discussion of the RATIO control. (Page 9) By shifting the input GAIN upward, further compression and improved signal to noise can be achieved, at the expense of dynamic range.

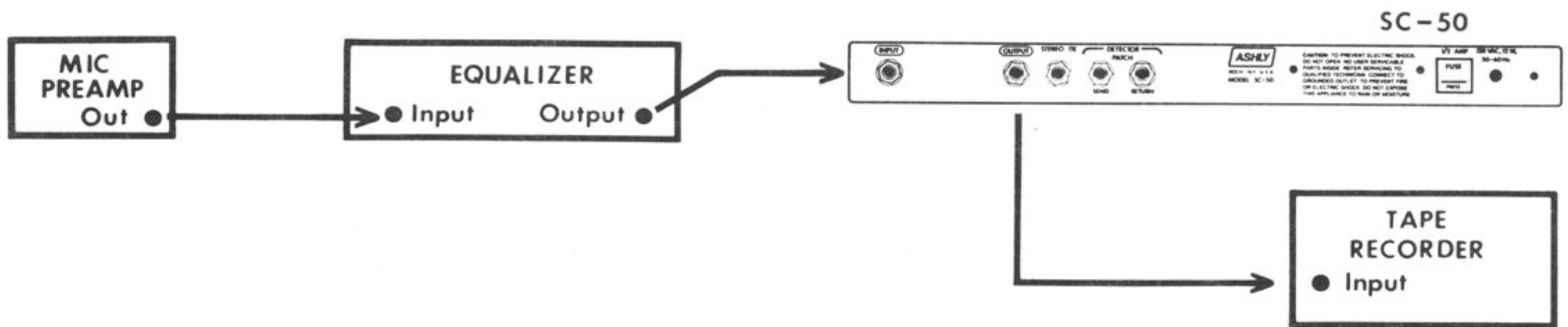


Figure 26 The SC-50 used to prevent tape saturation and improve S/N ratio.

DE-ESSING

A special type of saturation problem often encountered in recording is the sibilant (Ssss) sound of the human voice. High frequency, sibilant sounds can reach very high energy levels, such that a voice that is otherwise undistorted breaks up on the esses, producing a raspy, undesirable sound. With the current trend toward crisp, bright equalization of vocal tracks, the problem is magnified. Add to that the inherent tendency of magnetic tape to saturate earlier at high frequencies and the internal high-frequency boost (record pre-emphasis) of a standard tape recorder, and the need to control sibilants becomes apparent.

The solution is frequency-dependent limiting, which is easily accomplished with the SC-50. By inserting an equalizer into the Detector Patch points, as shown in Fig. 27, and boosting the equalizer at high frequencies in the

vicinity of the sibilant, the limiter's detector circuit becomes more sensitive to this particular range of frequencies, and so will limit the bothersome sibilants more than other frequencies.

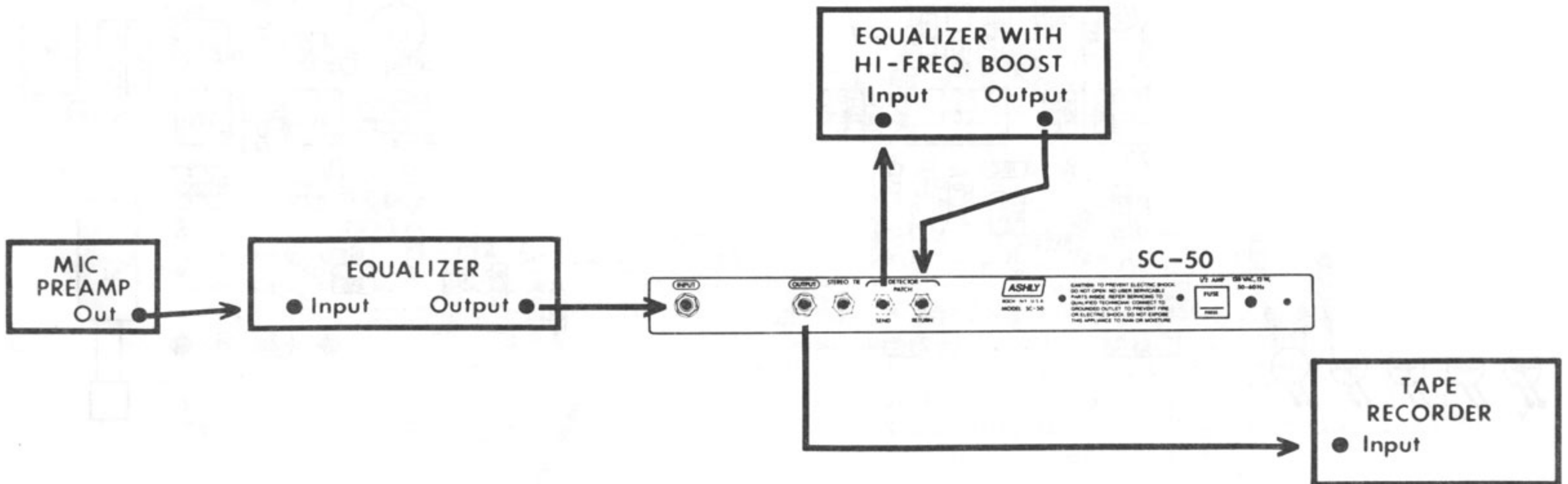


Figure 27 "De-Essing" with the SC-50.

Realize that this technique is very different from simple equalization. Equalizing a sibilant vocal by cutting high frequencies would result in a loss of important high frequency information at all times, whereas de-essing has no effect whatsoever on the signal except at the instant of the sibilant. At that moment, the SC-50 will reduce overall gain. Frequency response is unaffected, and the sibilant is controlled.

FM PRE-EMPHASIS MODIFICATION

When used as a limiter in conjunction with transmitters having standard 75 microsecond pre-emphasis, the detector circuit should be modified to lower its high frequency threshold. (See Figs. 28 and 29 for the SC-50 and SC-55, respectively.)

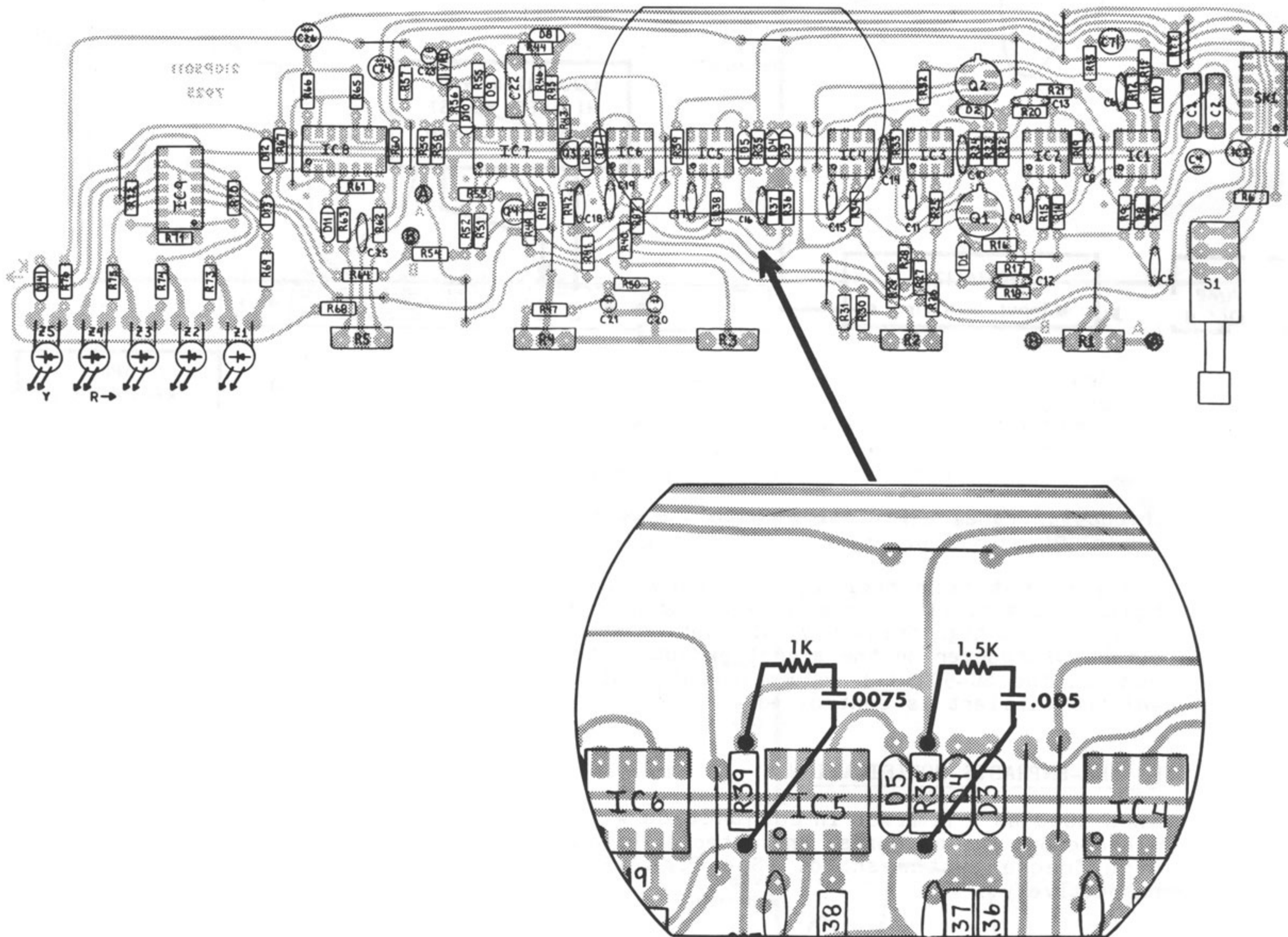


Figure 28 FM Pre-emphasis modification, SC-50. Capacitor values shown are for 75 microsecond pre-emphasis. For 25 microsecond values (Dolby), see the table below.

Note that no components are removed from the circuit. This illustration views the circuit board from the component side, although it is easiest to attach the new components to the copper side of the board. If you have further questions, consult the factory.

75 microsecond	25 microsecond
.0075 Mfd.	.0025 Mfd.
.005 Mfd.	.0017 Mfd.

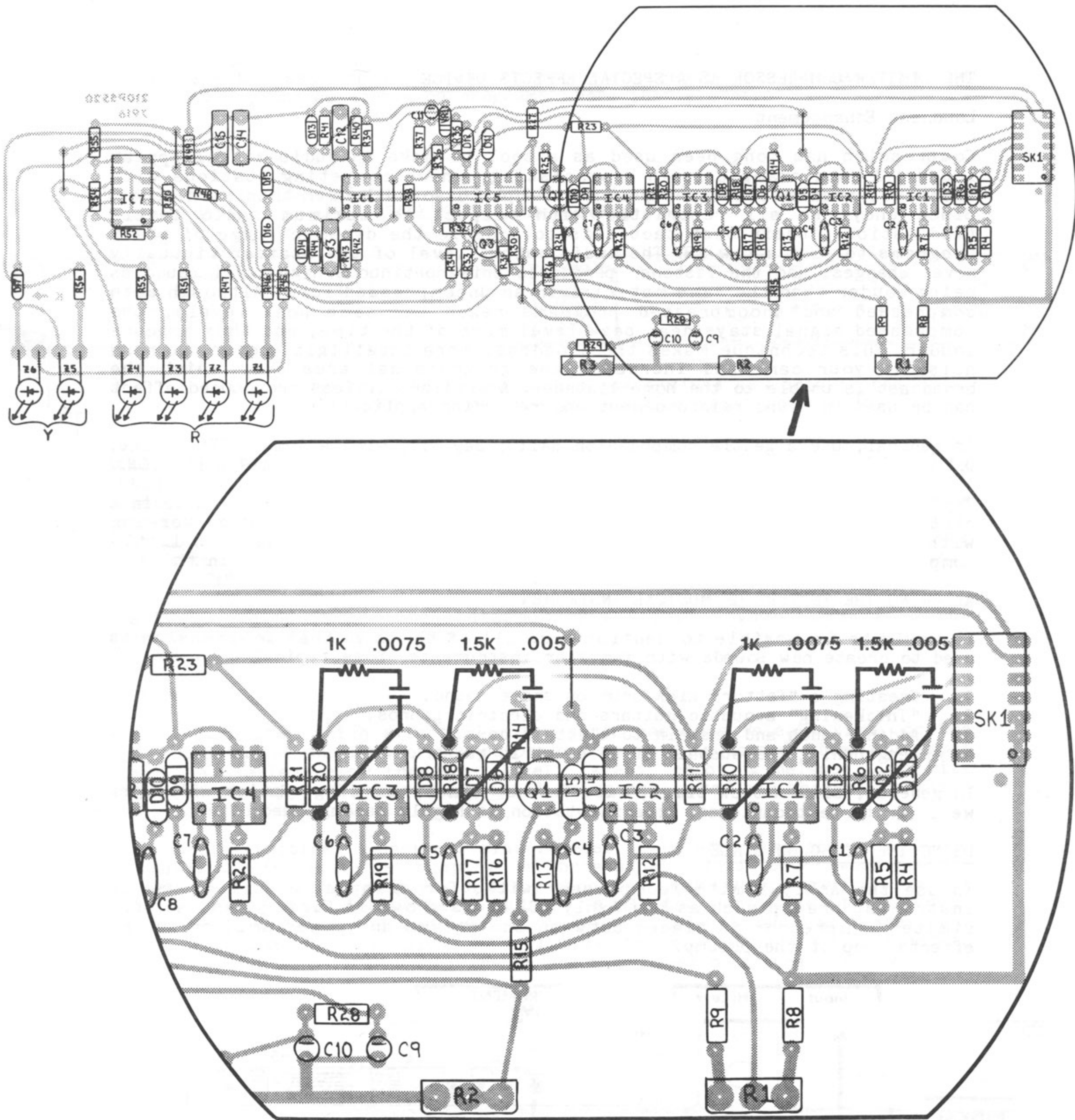


Figure 29 SC-55 FM Pre-emphasis modification.

*See notes on previous page.

THE LIMITER/COMPRESSOR AS A SPECIAL EFFECTS DEVICE

Loudness Enhancement

Compression has long been used as a tool to make an audio signal appear louder. A good example is in broadcasting, where competing stations with identical transmitters and power attempt to sound louder than each other. Since they are all restricted with respect to maximum audio levels (modulation), their best tactic is to squeeze the dynamic range of their programs to just a few dB. The audio output level of the station virtually never changes, and the listener perceives this continuous high-level sound as being louder than the same material in an uncompressed form. Although both compressed and uncompressed programs reach the same peak levels, the compressed signal stays near peak level more of the time, and thus appears louder. This technique makes the broadcast more intelligible over the road noise in your car, and increases the geographical area over which the broadcast is usable to the home listener. A similar, if less pronounced effect can be used in sound reinforcement and recording applications.

In general, use a gentle compression RATIO, say 4:1, with a 10ms. ATTACK time, 0.1 sec. RELEASE time, and enough GAIN to cause 6 to 10 dB of GAIN REDUCTION.

Try using this effect to help bring out a lead vocal or instrumental solo in a cluttered mix. The compressor is also a great corrective tool when working with singers whose own dynamic control is less than adequate. A little compression helps to keep their quieter lines from becoming buried in the mix.

ALTERING THE TEXTURE OF MUSICAL INSTRUMENTS

It would be impossible to mention here all of the ways that compression is used to create new sounds with familiar instruments. Some typical uses are:

1. Creating a "fatter" kick drum or snare sound.
2. "Thickening" acoustic guitars and electric pianos.
3. Adding punch and sustain to electric bass.
4. Lengthening the sustain of an electric guitar.

In general, slow Attack times, fast Release times, and large Ratios will work well for these techniques. Experimentation is highly recommended.

USING THE SC-50 ON STAGE

To use the Ashly limiter/compressor with, for example, a guitar, use an instrument preamp such as the Ashly SC-40. As shown in Figs. 30 and 31, the limiter/compressor is placed between the preamp and power amp, or in the effects loop of the preamp.

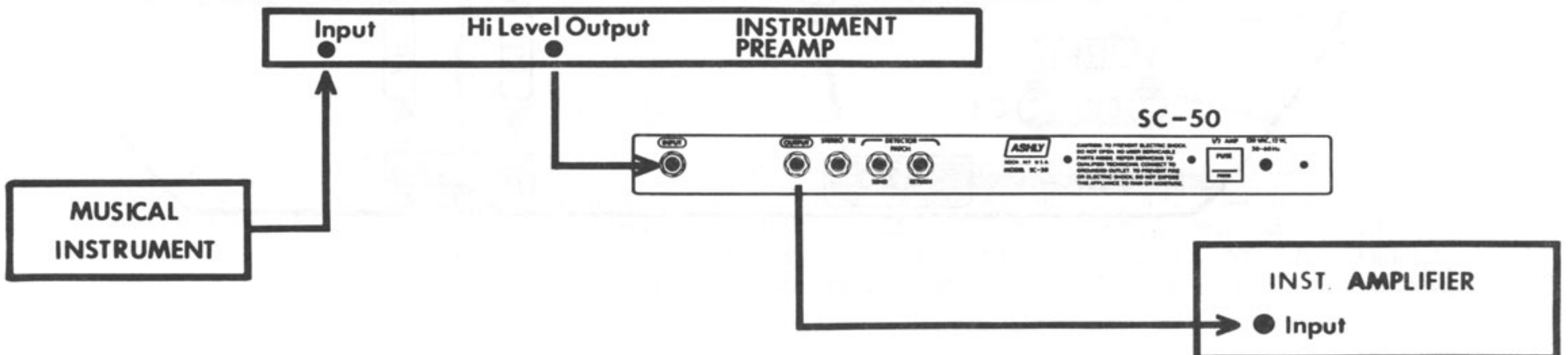


Figure 30 The SC-50 used with a musical instrument preamplifier.

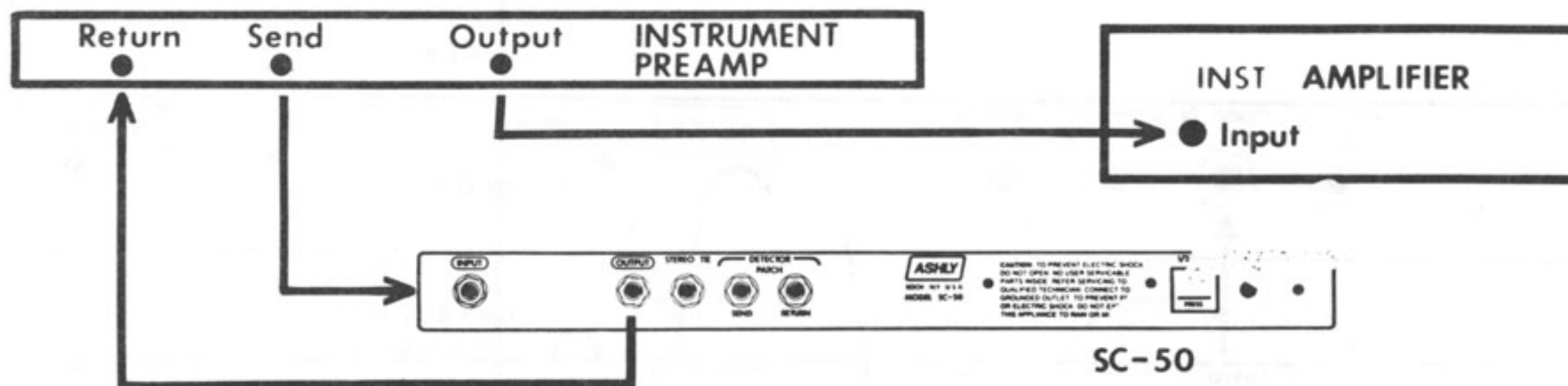


Figure 31 The SC-50 inserted into the effects loop of a musical instrument preamplifier.

VOICE-OVER COMPRESSION ("DUCKING")

The SC-50 can be used to automatically reduce music to a background level when an announcer is speaking. (Fig. 32)

In this scheme, only the music signal is actually gain-reduced by the SC-50. However, the Detector is connected to respond to an announcer's voice instead of the music's peaks. The music output and the announcer's voice are then combined in an external mix circuit. A variable gain control on the announcer's voice level is helpful in matching the announcer signal level to the nominal 0 dBV threshold level of the detector.

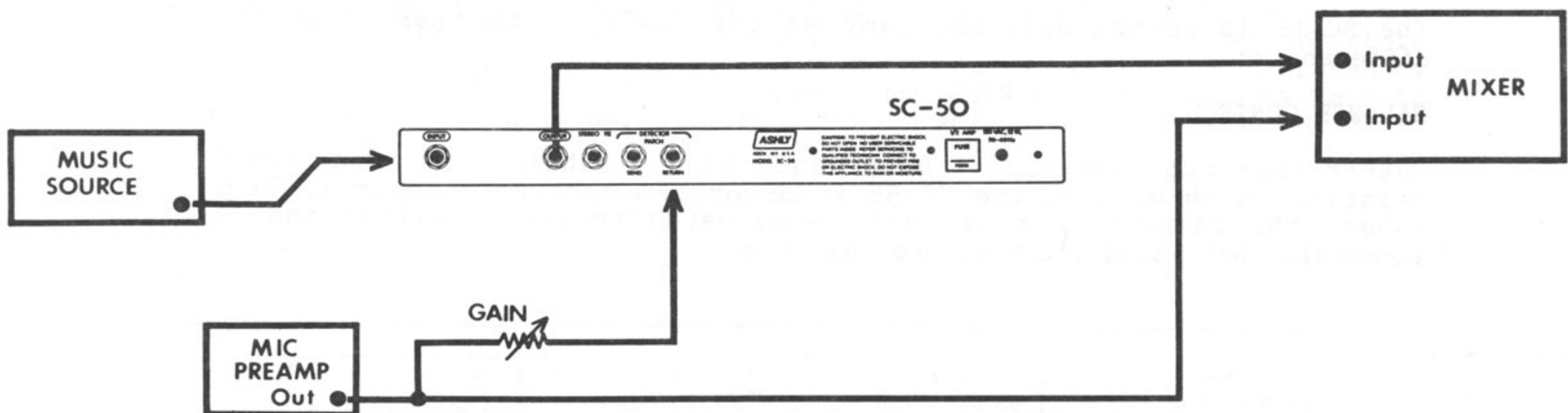


Figure 32 The SC-50 as a "Voice-Over" Compressor.

STEREO OPERATION

Two or more SC-50's may be tied together to provide accurate stereo limiting and compression. As shown in Figure 33, this is simply a matter of inserting a single patch cord between the STEREO TIE points of the two units. Notice that the GAIN REDUCTION LED's will still read independently, with each indicator corresponding to half of the total gain reduction action. The actual gain change is always identical for each limiter and is determined by the louder channel. In this way, there is never any shifting of the stereo perspective.

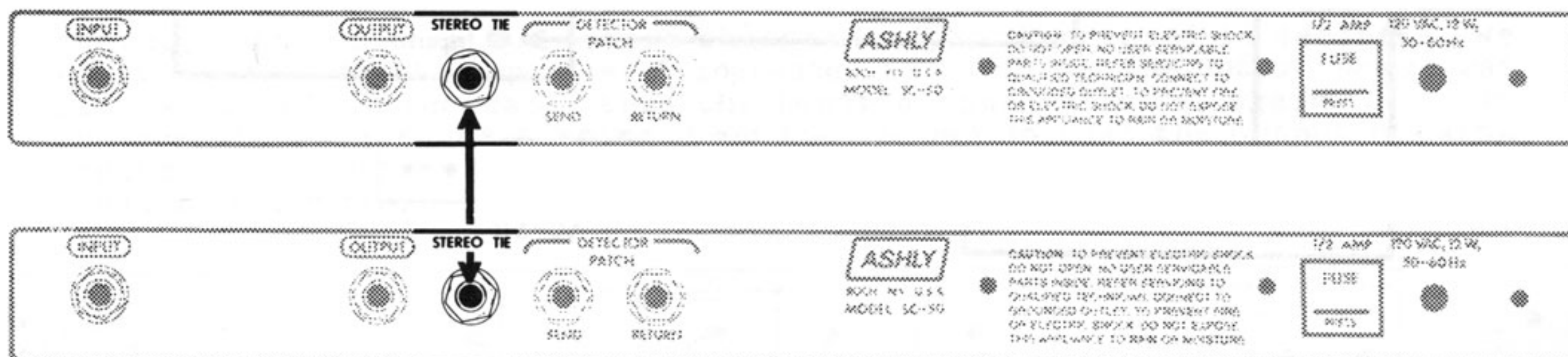


Figure 33 Linking two SC-50's for stereo operation is simply a matter of inserting a standard patch cord between the Stereo Tie patch points.

In permanent rack-mount installations, the Stereo Tie points of two SC-50's may be brought out to a convenient front panel switch (SPST) inserted into the "hot" line between the two patch points. This switch will then select either stereo or independent tracking of the two units.

SPECIAL NOTES ON THE SC-55 STEREO LIMITER/COMPRESSOR

The SC-55 is essentially the same as the SC-50, with these operational differences:

BALANCE CONTROL

Rather than two independent GAIN controls for setting the input level in relation to threshold, the SC-55 incorporates a BALANCE control. In most cases, the BALANCE control will be adjusted so that the Left and Right Threshold indicators light at the same time.

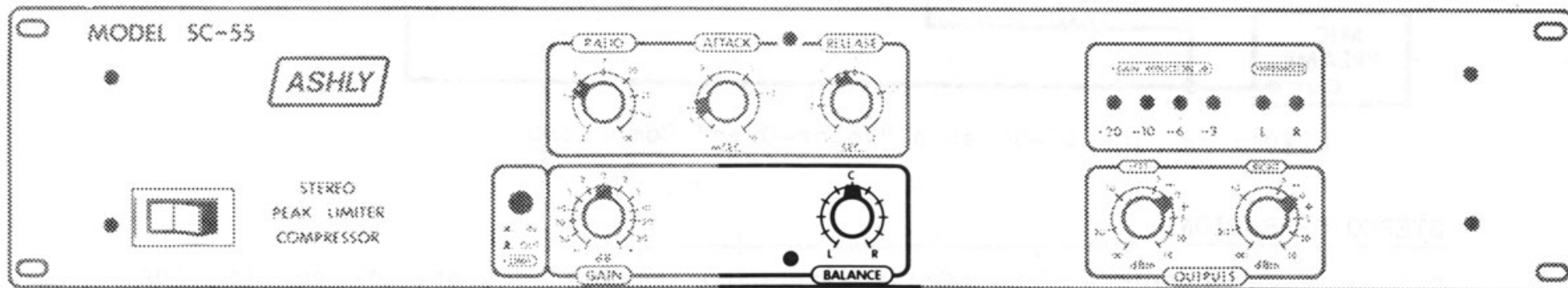


Figure 34 A BALANCE control is used rather than two GAIN controls.

GAIN REDUCTION/THRESHOLD INDICATORS

Since both channels of the SC-55 reduce gain by the same amount at all times, only one GAIN REDUCTION display is required. However, since either channel can achieve threshold first, two THRESHOLD indicators are used.

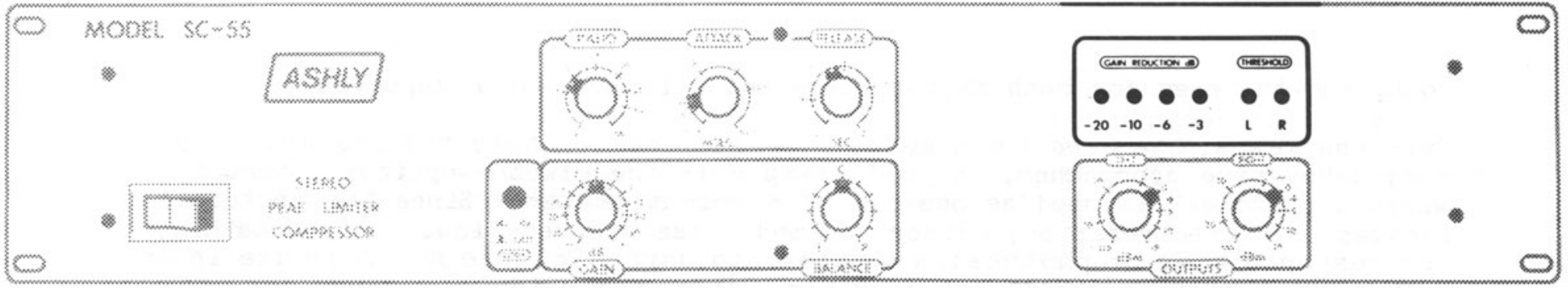
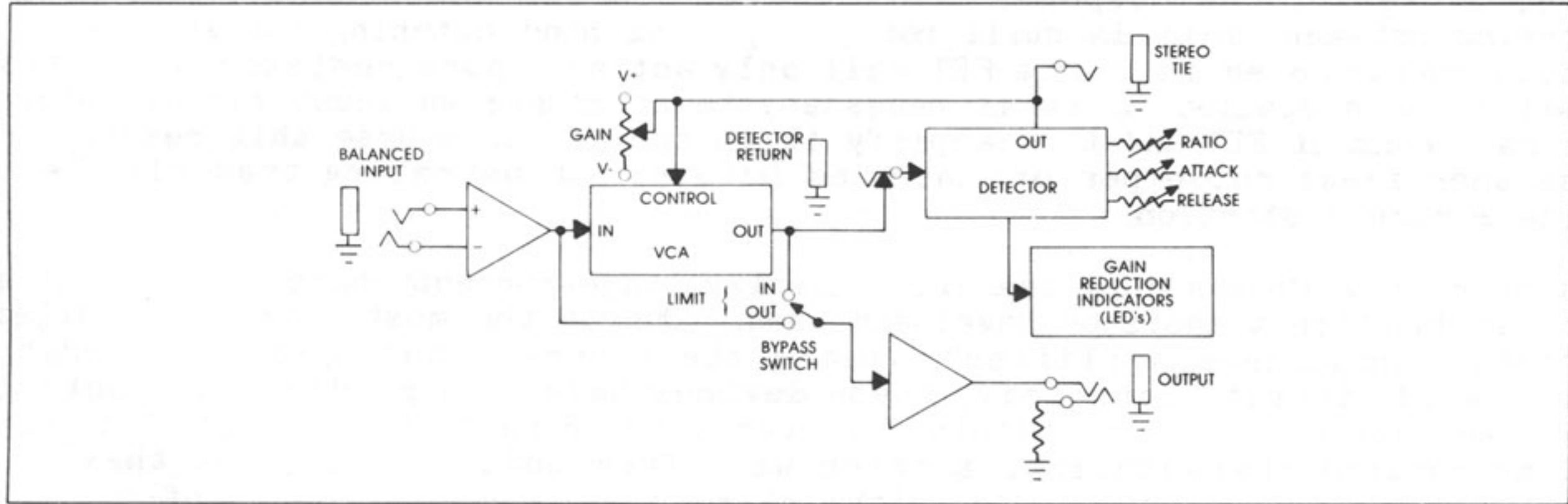
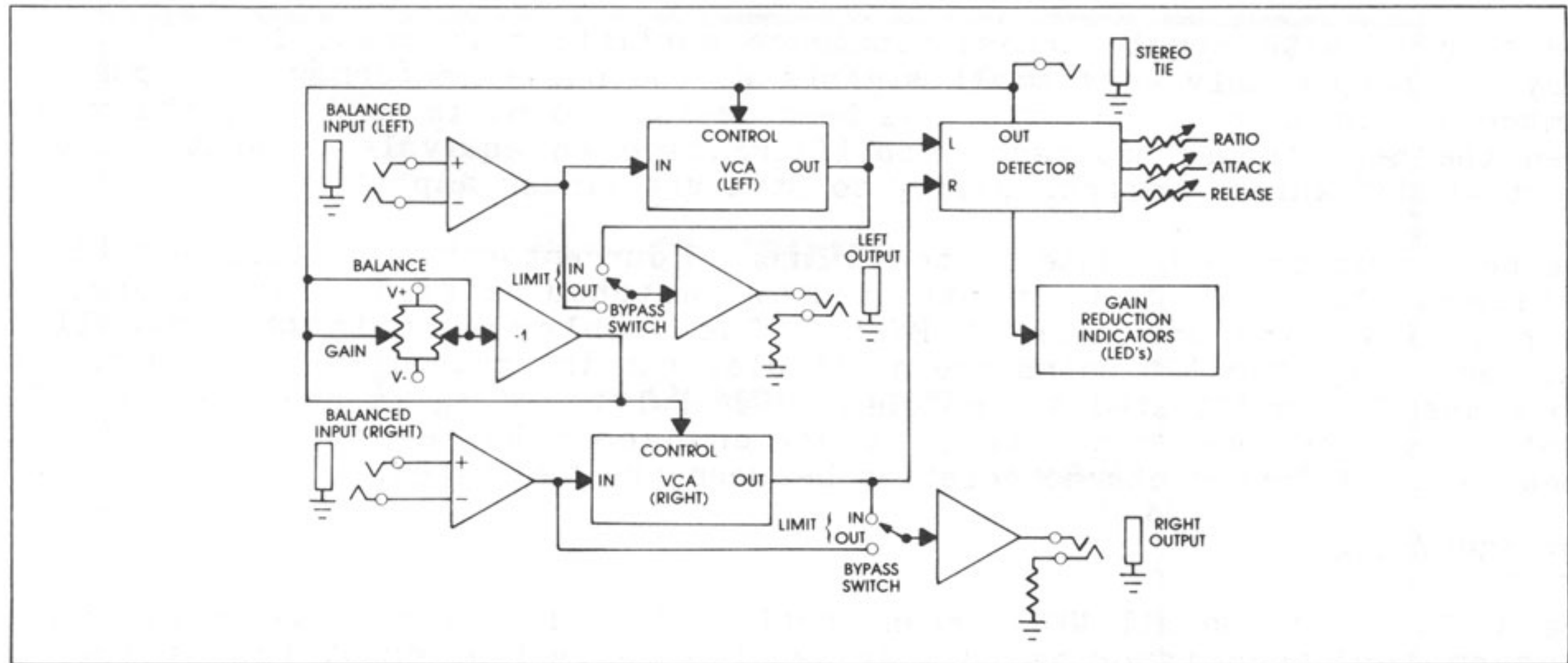


Figure 35 Two THRESHOLD indicators and a common GAIN REDUCTION display are used on the SC-55.



SC-50 BLOCK DIAGRAM



SC-55 BLOCK DIAGRAM

LIMITER/COMPRESSOR CIRCUITRY

Voltage Controlled Amplifiers

Early VCA's were based on vacuum tubes with a "remote cutoff" characteristic. The tube would simply change its gain in response to a changing bias voltage. Tubes developed for this purpose did an excellent job, in fact they could exceed the noise and distortion performance of today's best solid state VCA's. Unfortunately, they also had some serious disadvantages peculiar to tubes--change of gain and matching as aging took place, heat, microphonics, high

cost, and the need for both high-voltage and filament power supplies.

Over the years, the need for a good, low-cost, solid state VCA brought about many innovative approaches. A good example is the electro-optical attenuator where a photocell is used as one leg of a potentiometer. Since the photocell behaves as a true resistor, distortion and noise are very low. Unfortunately, the response time of photocells is slow and unpredictable so their use in a fast peak limiter is really not feasible. Also, the matching between units is very poor so that stereo tracking is not possible without tedious hand matching of photocells.

Another approach uses a field-effect transistor (FET) as a variable resistor. Here, at least, the response time is fast (in the nanosecond range), but matching between units is still poor, requiring hand matching for stereo. An additional problem is that a FET will only act as a pure resistor with very small signals applied so it is necessary to attenuate an input signal before the gain control FET and then amplify it up again. Of course this results in less than ideal noise performance and imposes a frustrating tradeoff: less noise = more distortion.

A number of VCA's based on the exponential voltage-current characteristic of a bipolar junction transistor have been used. One of the most common is called a "transconductance amplifier". Using the inherent matching obtained by integrated circuit technology, these devices have very predictable control characteristics. Tracking within 1dB over a 40 dB range is common. Not only do the control characteristics match well from unit to unit, but they can easily be made exponential (logarithmic) so that even increments of control voltage produce even increments of gain change in decibels. The response time is also very fast.

The problem with simple transconductance amplifiers is that, like FET VCA's, they can handle only very small signals so the noise performance is poor. A number of linearizing circuits have been devised to minimize this problem, but even the best transconductance amplifiers have an equivalent input noise of about -80dBV which compares poorly to straight linear amplifiers.

The best compromise to date is the "class AB current ratio multiplier." Early implementations of this circuit used two matched pairs of transistors, one pair of NPN's and one pair of PNP's. The problem here is that excellent matched integrated NPN pairs are available, but integrated PNP's are not. The PNP's must be hand tested and matched. Careful trimming is necessary for low distortion and even minor temperature changes make re-trimming necessary because of differing characteristics between the two types.

THE ASHLY VCA

The Ashly VCA is an all NPN current ratio multiplier circuit using integrated "supermatch" transistor pairs. It has low noise (-90 dBV), low distortion (.05%), excellent response time and tracking and does not suffer from thermal drift. It is a little complex - there are 5 supporting op-amps involved, but the low cost of good op-amps these days makes this feasible. While the noise and distortion are still not equal to the best linear amplifiers, they are at state-of-the-art levels and the circuit is consistent in mass production with no trimming or hand selection of transistors.

DETECTORS

It would seem that, of the two components in a limiter/compressor, the VCA is the more critical since the audio passes through it and the detector only provides it with a control voltage. Experience showed us that both are crucial to the overall sound and that, if anything, the detector's performance is the harder to judge by conventional measurement techniques. While the VCA is doing

its job if it has low noise and distortion, the detector must constantly adjust the gain of the audio path in a manner which keeps the level under control while sounding acceptable to the listener. This constantly changing gain is a DYNAMIC action, and conventional audio measurements like noise and distortion checks are STATIC (constant level). We became painfully aware of this problem with some of our earlier limiter prototypes which measured fine and sounded terrible. This led us to use a purely subjective approach in the design of the detector; we did a lot of listening to determine what sounded good and what didn't.

Two important features emerged from this research:

1. We designed the detector to let the attack and release times speed up as more and more limiting occurs. The compression ratio also increases. This lets us maintain peaks fairly close to a constant ceiling level, but allows the illusion of increasing loudness as input level increases, thereby preventing complete loss of dynamics when limiting.
2. We incorporated a double release time constant. When release time was set slow with a single time constant, transients such as mic "pops" and record scratches caused a quick reduction in gain and a slow fade-up, making the action of the limiter very obvious. With the double time constant, release from gain reduction after a brief transient is always fast, with a slower release after a sustained overdrive.

When choosing a limiter/compressor, you can see that it is very important to listen to it in your particular application and see that it sounds the way you want. There are lots of these devices with seemingly excellent specs which sound very different with real program material applied to them.

PEAK OR RMS... AN OFTEN OVERLOOKED AND MISUNDERSTOOD DIFFERENCE

There are several ways of looking at a signal to determine its level. A peak detector looks at the maximum voltage a signal reaches regardless of its waveform, while an RMS (root mean square) detector looks at the energy in a signal regardless of the short term voltage levels. This makes a peak detector the correct choice for preventing clipping, overmodulation, or tape saturation, while an RMS detector is used to restrict material to a given loudness. When an RMS limiter is used to prevent clipping, the result is unpredictable. For instance, a flute and a snare drum which are limited to the same RMS level might have peak levels as much as 30 dB apart! Use peak limiters to prevent clipping.

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.

ATTACK TIME

The amount of time that elapses before a limiter/compressor begins to attenuate the output level after threshold has been reached.

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.

BREATHING

A usually undesirable fluctuation of background noise resulting from compressor action. (Also called "Pumping.")

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.

COMPRESSOR

An amplifier which reduces its gain as its input is increased beyond a predetermined "threshold."

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.

EQUALIZATION

Modification of the frequency response of an audio system for either corrective or enhancement purposes.

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 self stabilizing, while positive feedback causes instability.

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.

GAIN REDUCTION

The amount (expressed in dB) by which a limiter/compressor's output has been reduced in level with respect to its uncompressed level.

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.

LIMITER

An amplifier which reduces its gain as its input is increased beyond a predetermined threshold. Usually used to protect audio systems against sudden, high level signals, and possible overload.

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.

PREAMPLIFIER

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

RATIO

The relationship between change in input level and resultant change in output level.

RELEASE TIME

The time required for a limiter/compressor to restore system gain to normal after the input signal has fallen below threshold.

SATURATION

The point at which the magnetic storage capability of a piece of recording tape is exceeded.

SIBILANCE

The distortion caused by loud high frequency signals, such as the "Ssss..." sounds in human speech.

THRESHOLD

The level above which a limiter/compressor begins to reduce gain.

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.

TROUBLE SHOOTING TIPS

NOTE: Unshielded cables, improperly wired connectors, and cable with broken strands of wire rattling around are very common problems. Use good quality cables with good quality, correctly wired connectors.

NO OUTPUT

Check AC Power. Is the pilot light on? Check input and output connections--are they reversed? Are you sure you have an input signal?

CONTROLS HAVE NO EFFECT

Is the LIMIT IN/OUT switch IN? Perhaps the RATIO control is set too low to produce an audible effect (see introductory notes on Applications, page 16), or the input level is below threshold. Is the THRESHOLD LED lighting up? If not, increase the GAIN control. Do not expect to hear any effect when the input level to the SC-50 is below threshold, since the unit is simply a linear amplifier at those levels.

WHEN USING HEAVY COMPRESSION, BACKGROUND NOISE IS NOTICEABLE DURING QUIET SECTIONS OF THE PROGRAM

As defined in the section on Compression (page 13), quiet program material is effectively made louder while loud peaks are made quieter. When the program source is thus raised in volume, its noise floor is also raised in volume by a proportionate amount. This is not a defect in the limiter-compressor, but an unavoidable side effect of the gain altering process. If the noise becomes a problem, the solutions are to either: a) clean up the program source, or b) use less compression.

EXCESSIVE HUM OR NOISE

Hum is often caused by a "ground loop" between components. Try using the suggested balanced input and output hookups if the other pieces of equipment used in conjunction with the SC-50 have balanced inputs and outputs. See page 5. Noise can also be caused by insufficient drive levels. Make sure you are sending a nominal 0 dBV line level signal to the unit.

If you are still having problems, get in touch with your Ashly dealer, or call the factory direct, toll free at (800) 828-6308. In New York State, phone (716) 544-5191.

SPECIFICATIONS

SC-50

CONTROLS
defeat switch
gain $\pm 30\text{dB}$
ratio 2:1 - ∞
attack time 200 μs - 20 mS.
release time 100 mS - 2S
INPUT IMPEDANCE 10K Ω balanced bridging
OUTPUT IMPEDANCE 50 Ω term with 600 Ω
or more
DETECTOR PATCH POINT allows the connection of an equalizer in the detector loop to produce frequency selective limiting.

MAX. IN-OUT LEVEL +20dBm
FREQUENCY RESPONSE $\pm .5\text{dB}$ 20Hz-20kHz
DISTORTION < .05% THD, 0dBV, 20Hz-20kHz, no limiting
< .2% +18dBV worst case.
HUM AND NOISE -90dBV, unity gain
POWER 120 VAC, 50-60Hz, 5W
SIZE 19"L x 1 3/4"H x 6"D
SHIPPING WEIGHT 8 lbs.

SC-55

CONTROLS
defeat switch
gain $\pm 30\text{dB}$
ratio 2:1 - ∞
attack time 200 μs - 20 mS.
release time 100 mS - 2S
INPUT IMPEDANCE 10K Ω balanced bridging
OUTPUT IMPEDANCE 50 Ω term with 600 Ω
or more
MAX. IN-OUT LEVEL +20dBm

FREQUENCY RESPONSE $\pm .5\text{dB}$ 20Hz-20kHz
DISTORTION < .05% THD, 0dBV, 20Hz-20kHz, no limiting
< .2% +18dBV worst case.
HUM AND NOISE -90dBV, unity gain
POWER 120 VAC, 50-60Hz, 5W
SIZE 19"L x 3 1/2"H x 6"D
SHIPPING WEIGHT 10 lbs.