Technical Notes Vol. 2, No. 1A

1/3 Octave Equalization and The JBL/UREI 5547A and 5549A

Introduction:

The use of one-third octave graphic equalizers in the audio world has expanded rapidly in the last several years. Along with the expanded use have come a wide variety of different models from many manufacturers. While they may appear similar on casual inspection, they do in fact differ. It is the purpose of this paper to describe some of the important design and performance considerations which are common to all graphic equalizers and to discuss the performance advantages available from the new JBL/UREI Model 5547A Graphic and Model 5549A Room Equalizers. This discussion is not a mathematical treatment of the subject. It will, however provide a better understanding of the subject to those who neither have nor need an extensive background in filter theory.

Filter Shape and Combining Action:

Any discussion of the important parameters of a one-third octave equalizer must start with considerations of filter combining action, filter shape and minimum/non-minimum phase behavior. When we talk about filters which combine we are actually talking about two different aspects of the design. The first is the method by which all of the filters are connected together. In the mathematical sense combining filters are those which multiply while non-combining filters add. Or, to put it another way, combining filters add decibels – non-combining filters add volts. To illustrate why combining is desirable, let us take as an example a pair of filters which, because of their filter shape, happen to cover a certain band of frequencies in common (see Fig 1).

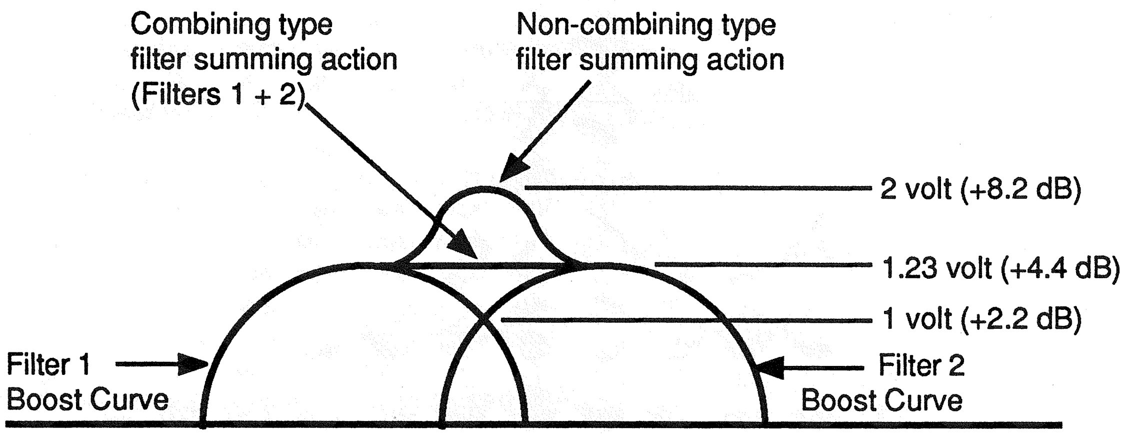
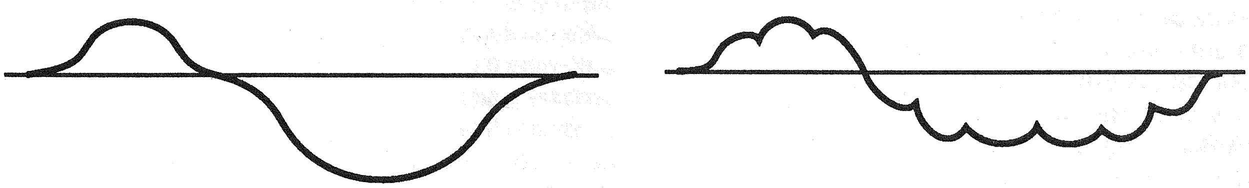


Figure 1. **Summing Action of Combining and Non-Combining Type Filters**

Let us say that one of the filters is adjusted in such a manner that the level of signal at one frequency in this common band is raised to 1 volt rms (approximately +2 dBu.) The control position is marked and the control returned to zero. The second filter is then adjusted to make the signal level 1 volt rms at the same frequency. Now the first filter is returned to the previously marked position. What is the output level of the equalizer at that frequency? If the filter is combining, it will add the two levels in dB and get +2 plus +2 = +4 dBu (approximately 1.23 volts rms). If the equalizer is non-combining, it will add the two voltages together and get 1+1 = 2 volts rms (approximately +8 dBu). Rather a difference! A one-third octave audio equalizer should have filters which add decibels for smooth, predictable combining action of multiple filter sections.

The second aspect of a good combining filter is that the filter shapes should be designed to achieve the smoothest, most ripple-free amplitude response over the widest range of control settings possible (See Fig 2). There is always a tradeoff between selectivity (isolation between filter sections) and smooth amplitude response. It is possible to design a filter set with a wide range of Q's and variation of Q with control setting. The Q of a one-third octave filter is theoretically 4.3, but few, if any, commercial units actually measure 4.3 except at one setting of the filter. This is because it has been found that to achieve good, low ripple combining action, the Q of the filter needs to change with the amount of boost or cut so as to smoothly blend with adjacent filter sections (see Fig 3). The result is that the filter shape is broader at small amounts of Boost/Cut, and becomes increasingly more narrow with greater amounts of Boost/Cut. This creates minimum amplitude response ripple for a wide range of control settings. The low amount of amplitude response ripple keeps the phase variation low.

At least one manufacturer of one-third octave equalizers is making a strong sales pitch for their units based on the fact that because of the way their units are designed there is little, if any, change in the shape of their filter curves at any position of the Boost/Cut controls. This is an interesting design choice, but we do not believe it to be well considered. While the design does offer increased selectivity at the one-third octave center frequencies, it simultaneously introduces a greater amount of phase shift and increased amplitude response ripple



A. Good Combining Action **B. Poor Combining Action**

Figure 2. **Equalization curves with Good Poor Combining Action**

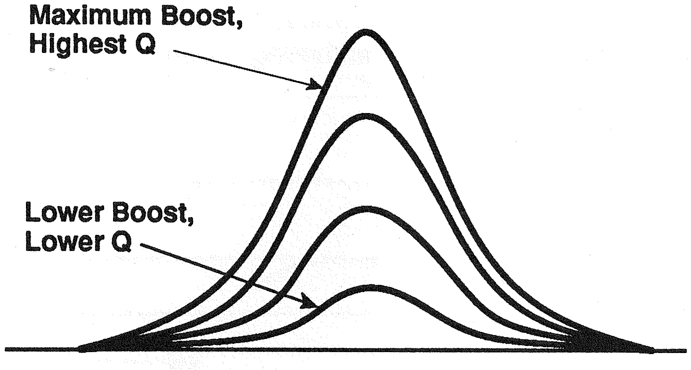


Figure 3. **Q Changes as a Function of Boost/Cut for Low Ripple and Good Combining Between Filters**

for any setting of the controls.This runs directly counter to accepted theories of what sounds best; a frequency response correcting device should not introduce additional amplitude and phase response errors into the signal path. There may be some confusion on the part of the designers of that device and also for many users who do not understand that one-third octave devices are still broad-band devices and should not be used to perform the functions of a narrow-band device. Narrow band filters such as the UREI Model 562 are more suited to the control of feedback with minimal disruption of the amplitude and phase response of the system. In addition, feedback "room modes" do not necessarily occur on the exact center frequencies of the one-third octave equalizer. The ability to adjust adjacent filter sections of a good combining filter so that the apparent center frequency of the equalizer is between the ISO center frequencies allows for smooth combining action for any response adjustment. Good combining action is clearly preferable to increased amplitude response ripple and the resulting phase shift.

Minimum and Non-Minimum Phase:

The correct type of filter for use in an audio system is referred to as minimum phase. This means that the equalizer produces only the minimum amount of phase shift as determined by the amplitude response variation. There is a class of filters which has this characteristic. The definition of that class of filters is a mathematical statement about the structure of the filters which very strictly limits their design. (In mathematical terms, the filters do not contain poles or zeroes in the right half-plane of the LaPlace transform, and the log magnitude and phase are related through the Hilbert transform). It is the goal of all electronic components incorporated in an audio system, including loudspeakers and microphones, that they approach minimum phase response.

The earliest one-third octave filters were not designed for listening to audio; rather, they were designed by manufacturers such as General Radio and Bruel & Kjaer for use in laboratories to aid in the study of vibration phenomena and the like. For the scientific uses envisioned by the designers of those filter sets a high amount of isolation was necessary between filter sections. The filter shapes were basically rectangular and the frequency response of this type of filter exhibits a stepwise shape. A filter of this sort is non-minimum phase. This is because of the high amount of phase shift inherent in the design of filters able to achieve this high degree of isolation.

When early attempts were made to use these filters in audio systems the results did not sound as good as had been hoped for. The sound was described as sounding like someone was talking in a barrel. But, despite the all too obvious deficiencies of these filters, it was apparent that they were capable of smoothing the amplitude response of a sound system, and that in doing so worked in the direction of improving both the intelligibility and the naturalness of sound reinforcement systems. The problem was getting rid of the barrel-like sound caused by the tremendous amount of phase shift between adjacent filter sections. The solution was the use of minimum phase combining type filters. Minimum phase does not mean zero phase shift. In all naturally-produced filters, phase shift is an unavoidable byproduct of any amplitude response variation. Although it is possible to synthesize filters with zero phase shift by using all-pass filters or time delays, such filters will then adjust only amplitude response, leaving the corresponding naturally produced phase shift uncorrected. This is, of course, undesirable.

In corrective equalization it is important to remember that there are two types of frequency response anomalies that may be present in a room – minimum phase and non-minimum phase. Non-minimum phase responses are due to the multiple paths that the sound takes from the source to the listener. The length of these paths will be different and the resulting time delays will form an acoustic transversal filter. The path length differences will cause partial or even complete cancellation of sounds at certain frequencies which have particular wavelength relationships to the path length differences. These frequencies at which cancellation occurs will be different in different areas of the acoustic environment as the path lengths change and as various reflecting surfaces become more or less important. It is unreasonable to attempt to correct for non-minimum phase response anomalies for two reasons: first they are not global: that is, they are not the same throughout the acoustic space. What is correct for one area will not be correct for another only a few feet away. Second, they may not be fixable. Imagine a situation in which the path lengths and amplitudes add up just right to produce a single 60 dB notch in the frequency response at 1000 Hertz. Now, imagine yourself trying to equalize the frequency response back to flat. If you were to find an equalizer with which you could dial in 60 dB of boost at exactly 1000 Hertz, what would happen? First you would find two very large bumps in your frequency response because cancellation notches of this type are generally quite narrow band, and the filter which you used to correct for it probably was not. Secondly, right in the middle of those two bumps is your 60 dB notch. It is still there because the path lengths and amplitudes which combined to

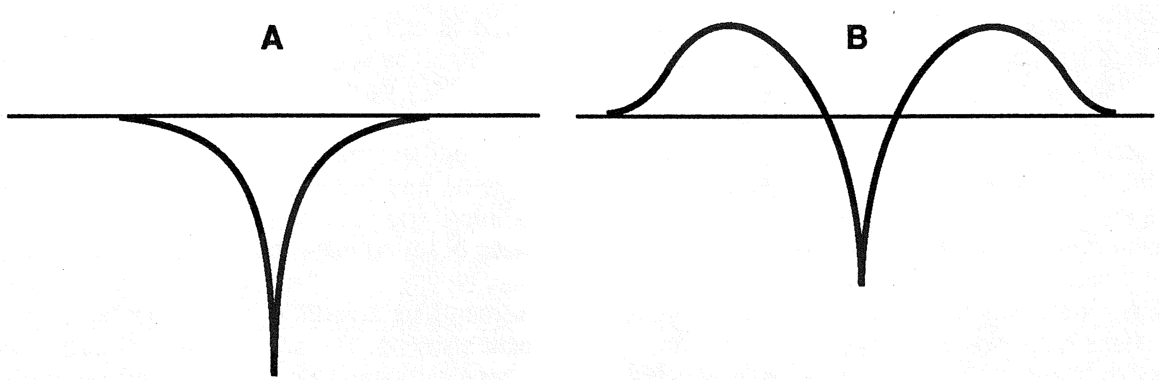


Figure 4. **Two Acoustic Signals Interact to Form a 60 dB Notch at 1 kHz (A), and What Happens When 60 dB of Boost at 1 kHz is Applied Using a 1/3-Octave EQ (B)**

produce it in the first place are still there. And, because for every amplitude response change you make you get a corresponding phase change, you now have undesirable phase shift of the worse sort. Oh, and by the way, you're probably in massive feedback if you have a microphone turned on.

This example is, of course, somewhat extreme. It is unlikely that only one reflecting surface would exist, and, to the extent that there are multiple reflections, each one does tend to fill in the "holes" caused by others until, at the opposite extreme for steady-state signals, all "holes" are filled in because of the extremely diffuse nature of the reverberant field. For the non-steady-state signal, none of this applies; the only solutions lie in the province of the acoustical designer. For small amounts of non-minimum phase caused response irregularity, it may be possible to correct the amplitude response, but it will not be possible to simultaneously correct for the phase response errors – in fact, correction of the amplitude response will probably increase the phase error.

There are also minimum phase response anomalies in room frequency response. These are caused by acoustic "filters" which modify the frequency characteristic of the sound reaching the listener. Some of these minimum phase "filters" include the low frequency response rolloff due to the size and mounting arrangement of loudspeaker enclosures, the high frequency response rolloff with distance due to excess attenuation of short wavelengths by air, and certain wideband response irregularities caused by the size and shape of the room and any other acoustical spaces which are coupled to it. It is possible to correct for both the amplitude and phase response anomalies in rooms which are minimum phase by using a minimum phase filter of inverse amplitude characteristics. Therefore we build filters to correct what we can correct, and leave that which we cannot correct to the acousticians.

Boost/Cut vs. Cut Only:

In one form or another, equalization of sound systems has been around since the thirties. However, it was not until Dr. Paul Boner's work in the early sixties that sound system equalization came of age. Later in that decade Altec introduced the hardware that began to make equalization a common practice among sound contractors. These early devices were passive loss networks capable of cut-only action. As active devices became available they initially imitated the cut-only action of the passive units. But soon manufacturers produced units which could boost as well as cut. The question then arose: "Which do I use – boost or cut?"

The answer is that, for room equalization, cut is best, but that with a knowledge of the limitations in its use, some small amount of boost may be acceptable. There are several reasons for the choice of cut-only EQ. First, of course is the realization that when performing the equalization there is a good deal of difficulty in determining which frequency response anomalies are amenable to correction. The natural tendency is to 'fill in the holes' of the frequency response with boost equalization. Unfortunately, if the cause of the dip is not amenable to correction as described earlier, then no amount of boost EQ will help. In addition, the effect of boost EQ is more easily heard by the ear and sounds less natural than an equalization curve arrived at by cut EQ. One individual has described the effect of boost equalization as similar to looking out over an empty field dotted with telephone poles. Cut EQ is then similar to the same open field except that the poles have been replaced by telephone pole size/shape holes. The poles are clearly visible, but the holes are not! The analogy is sound. The ear is much more sensitive to the effects of boost EQ than to the effects of cut EQ.

To some extent boost equalization may be used to smooth the amplitude response. The amount that may be used is determined by several factors including the type of system, the type of program material, the frequency at which the boost EQ would be used and the sensitivity of the listeners to the effects of the equalization. Generally, more critical program material, better listening environments, and more sensitive listeners are less able to tolerate boost equalization. In addition, the effects of boost equalization are more easily heard (to their detriment), when the frequencies being boosted are in the middle of the audio frequency band. The effect of excess boost will be heard as artificial and the filters may ring. The JBL/UREI Model 5549A Room Equalizer is recommended for the correction of room response anomalies. With 15 dB of cut available at each of 30 bands of one-third octave equalization and separate end cut filters it provides the range of control necessary to deal with the wide range of situations found in both fixed and portable sound systems of all types.

Creative Equalization:

The one-third octave graphic equalizer is obviously one of the many powerful tools in the repertoire of the creative audio mixer and its use is fairly well known. It offers the ability to shape the audio spectrum in an almost unlimited way while simultaneously presenting a front panel display of control settings that makes it very easy to understand at a glance what has been done to the EQ (and what yet needs to be done). The preceeding comments about filter shapes and combining action are of special importance to the achievement of a sound that is not only balanced in tone, but remains musical. At the same time it must be said that when it comes to creative equalization, just as with most things subjective, what works is right. For creative tasks we recommend the JBL/UREI Model 5547A Graphic Equalizer. With a boost/cut range of ±12 dB, 30 bands, and separate end cut filters, it incorporates both the high performance standards and the high degree of control flexibility demanded of a 'studio quality' audio product.

Inductors: Wire-Wound or Synthesized?

The most common method of designing one-third octave filters has been through the use of series L–C filters which offer better stability and lower sensitivity to component tolerances than most other filter types. In the design of a high quality L–C equalizer, therefore, one of the most important tasks is the design and construction of a series of high quality inductors, at least one for each filter section. Inductors have traditionally been made of magnet wire wound on a core of magnetic material. In recent years the wire-wound inductor has seen considerable competition from an electronic circuit called the synthetic inductor. UREI has manufactured one-third octave graphic equalizers using wire-wound inductors since 1972, and one octave band graphic equalizers using synthetic inductors since 1975. With this experience in the use of both technologies, we feel that we have a good understanding of the benefits and the limitations of both.

With good design and manufacture, wire-wound inductors may be made to a very high level of quality. As used here the term "quality" is concerned with the following performance factors:

1. Controlled, consistent Q of the inductor (not to be confused with the overall circuit Q.)
2. Consistent, precise inductance value.
3. Freedom from distortion caused by core saturation or other non-linear behavior.
4. Long-term reliability.

Parts which meet these criteria require good design, materials, manufacture and testing. The resulting wire-wound components do the job, but have several drawbacks: size and weight, particularly with respect to the low frequency coils; potential susceptibility to hum field pickup; and high cost.

The hum field pickup problem can largely be reduced by appropriate shielding, but at additional size, weight and cost penalties. The cost of the coils and of the associated precision capacitors needed to produce a high quality one-third octave band equalizer may approach 75% of the materials cost of the product. This has, and will continue to keep the price of such units out of reach of many users. If the product cost is to be reduced significantly the cost of the L–C networks must be reduced.

In recent years several manufacturers have developed one-third octave equalizers which use an electronic circuit to simulate the action of the inductor. As compared with using real inductors made of wire, the active inductor (also sometimes called a gyrator), at first glance, seems to offer significant advantages for use in a one-third octave equalizer. It is compact, light weight, immune to the effects of hum fields, the Q and inductance are controllable, and it offers significant cost savings over the traditional wire-wound inductor. Additionally the synthetic inductor uses components which are already designed for convenient printed circuit board mounting with no additional hardware.

The advantages of synthetic inductors must be balanced against their disadvantages. The first is their susceptibility to signal overload. In a series L–C circuit at resonance the voltage across the total circuit is multiplied by the Q of the circuit and this voltage appears across the inductor, as shown in figure 5. This means that if, for example, a signal of 1 volt is applied to an L–C network with a Q of 10 then 10 volts will appear across the inductor. When designing a wire-wound inductor to handle these voltage levels without saturation, it may be necessary to increase the size of the magnetic core and/or increase the number of turns of wire . At low frequencies this can make for physically large parts. The synthetic inductor, on the other hand, is not limited by core size, but is limited by power supply voltage. Traditional circuit implementations of the synthetic inductor have used operational amplifiers operating from ±15 to ±18 volt DC power supplies. If the amplifiers allow operation all the way to the power supply rails, this then corresponds to between 10.6 and 12.7 volts rms. In actual practice this will be somewhat less, depending on the specific op-amp used. The Q multiplication in a synthetic inductor, in effect, puts gain into the inductor amplifier and causes it to overload sooner than the other amplifiers in the signal chain. This overload is then coupled back into the signal path and appears as an audible 'glitch' in the signal waveform, as shown in figure 6.

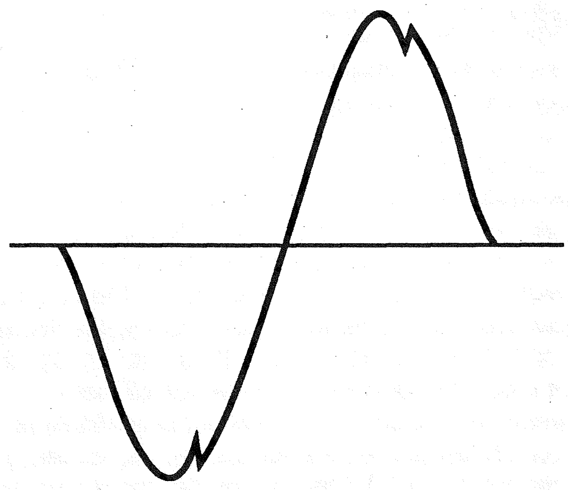
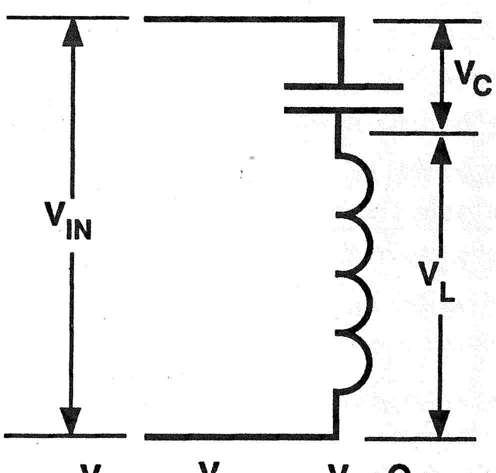


***Figure 6.* 'Glitch' in Signal Waveform Caused By Synthetic Inductor Overload**

Power supply voltage limitations are greater than they first appear. This is because the same voltage limitation applies for all of the synthetic inductors, not just some. Wire-wound inductors are more subject to core saturation at low frequencies (see Fig 7) and must therefore be very large to accommodate the overload requirements of professional equipment, but at mid and upper frequencies are capable of handling very large voltage swings with reasonable core size. The synthetic inductor, however, is subject to saturation at any frequency because of the power supply voltage limitation. The effect of Q multiplication on the synthetic inductor aggravates this effect.



$$V_L = V_C = V_{IN}\,Q$$

***Figure 5.* Voltage Across A Series L-C Network**

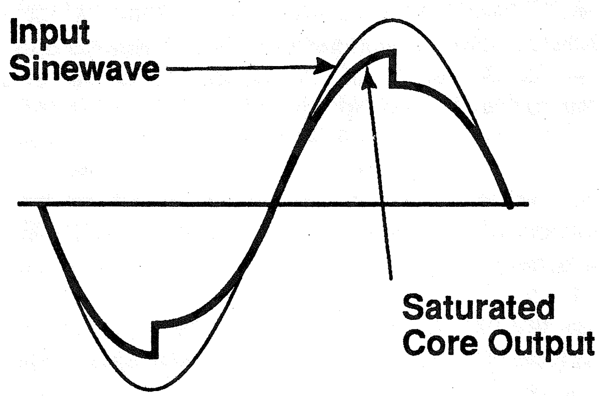


***Figure 7.* Core Saturation at Low Frequency in a Wire-Wound Inductor**

The likelihood of inductor saturation is not always apparent. It is easy to see that it is not reasonable to expect undistorted sound in a situation where a high input signal level is applied to a circuit and a large amount of boost is then applied. This problem can be somewhat alleviated by level detection circuits which give an overload indication to the operator, in effect telling him to turn down the signal. But such circuits do not warn of the same type of distortion when it occurs in the cut mode. This distortion occurs because of the 'gain' which is present only in the gyrator amplifiers. The gyrator thus overloads with less than the rated maximum input signal of the equalizer. Octave graphic equalizers do not have the same problem because the Q multiplication is not as great.

The second disadvantage with synthetic inductors is noise. Wire-wound inductors don't generate much noise of their own to be coupled into the circuit. But synthetic inductors, being made of transistor amplifiers, can and do contribute significant noise. Interestingly, this noise is not something that shows up on most manufacturers' data sheets. This is because noise specifications are typically shown with the equalization controls, set for flat response. In this setting of controls most, if not all, of the noise which may be generated by the filters is removed by the balanced circuit configuration of the equalizer. This is the optimum, or best-case, noise output, but is not generally achievable in real-world situations. The problem arises when various equalizer sections are set to positions other than flat; now any noise generated by the filter section is coupled into the audio path. Depending on the exact circuit and component implementation of the synthetic inductor, this will be more, much more, or ridiculously much more!!

If you stop to think about it, noise addition in a boost equalizer is a natural thing to expect. You are, after all, adding gain to the system. If there is any noise to begin with, boost equalization will amplify it. In a circuit which adds no noise of its own, the increase in noise due to boost will be in direct proportion to the amount of boost and to the bandwidth of the filter. Moreover, the additional noise will only be in the bandwidth covered by the filter. In a noisy circuit, the addition of noise will be greater than that to be expected from just the gain increase, and the noise bandwidth may be greater than the filter bandwidth.

Noise addition in a cut filter is not necessarily something that one might expect. But if the synthetic inductor circuit produces excess noise it will be coupled into the program material in the cut mode just as in the boost mode. Cut equalization is typically accomplished by forming a frequency sensitive attenuator in the signal path. Normally the shunt leg of an attenuator is taken to ground. In the synthetic inductor circuit, the op-amp output is the equivalent of ground. Unfortunately, if there is any noise at the output of the op-amp, then the supposed nice ground is an unwanted signal source. The result is that noise is injected into the signal path. The "gain" present in the synthetic inductor, due to Q multiplication, only makes the problem worse. The degradation of signal-to-noise ratio will be unexpected, and perhaps even more severe than in boost. Cut mode does, of course, pull the program signal down towards the noise floor. And if, while the program signal is being reduced, the noise is being increased, the signal-to-noise ratio is degraded very rapidly.

An Improved Synthetic Inductor:

Now that we have examined the problems that can occur with synthetic inductors, let us examine the circuit used in the JBL/UREI Models 5547A one-third octave Graphic and 5549A Room Equalizers. As it turns out, the easy way of making synthetic inductors using op-amps is really like using a cannon to kill a fly. Typical op-amps used in synthetic inductors have in excess of 20 transistors to give them all of the features which allow them to be used as general-purpose building blocks in modern analog circuit design. But the synthetic inductor doesn't need all of those bells and whistles. JBL/UREI engineering has designed a low-noise Class A transistor amplifier circuit and have packaged it as a hybrid microcircuit using the latest state of the art thick film, surface mount technology. The noise output of the JBL/UREI synthetic inductor is substantially lower than the best of the op-amp versions. In addition, the circuit is designed to operate from higher power supply voltages than an operational amplifier so that the maximum signal level it can handle is greater. The result is increased dynamic range under real world use.

What does this all mean to the user? It means that the typical specifications for overload and signal-to-noise on a graphic equalizer do not represent real world application of the products, and that without fairly extensive testing by the end user, there is no easy way to compare one unit to another. The signal-to-noise specification on an equalizer in the 'flat' position is only useful if you intend using the equalizer in the 'flat' mode. There is no way of telling what the noise might be at any position of the controls given only the noise specification of an equalizer set to the 'flat' position except by actual measurement. Additional specifications for noise measured with the filters in boost and cut are necessary to a clearer understanding of the differences between competitive units. And even then the differences may not be fully apparent due to differences in filter shape or combining action. About the only thing that would be safe to say is that most noise specifications for simulated-inductor one-third octave equalizers range somewhere between optimistic and misleading. Real-world usage will see a degraded dynamic range from that presented on the manufacturers' data sheets. The degree of degradation in the dynamic range will be a measure of the quality of the design and of the components used to implement that design. The design of the filter sections used in the JBL/UREI Model 5547A Graphic and 5549A Room Equalizers approach and in some cases exceed the performance characteristics of our graphics designed using wire-wound inductors.

Other Features:

The one-third octave filters are the most obvious area of interest in these equalizers. But considerable time and effort was spent in optimizing other areas of the product too. We will touch briefly on several of these areas and give the reasoning behind the design choices.

Headroom Control:

Signal-to-noise ratio is an important consideration when specifying a product for an audio system. But all too frequently much of the performance capability of a given piece of equipment may be thrown away during actual use. This is because the program signal must be kept below the maximum output capability of the unit to prevent clipping. The difference between the program level and the maximum undistorted output is called 'headroom' and will vary in different systems because of program material variations. For example, a large amount of headroom is normally required in a live recording studio situation where program level may change rapidly over a short period of time, but where compression or limiting is not yet appropriate (or even desirable). Conversely, an audio system which plays primarily pre-recorded program material with much more tightly controlled dynamic range may operate well with reduced headroom. In many systems, it is possible to determine the amount of headroom necessary for good operation. If, in such a situation, it were possible to optimize the signal level for each piece of equipment in the audio chain, it would be possible to make a trade-off of any excess headroom for improved signal-to-noise ratio. Unfortunately, this is not always easy to do.

An equalizer in a sound system should be operated at as high a signal level as possible to keep the signal-to-noise ratio high, but with 10 to 20 dB of headroom to account for the crest factor in the program material. These JBL/UREI equalizers are equipped with a novel and extremely convenient method for accomplishing headroom adjustment. To understand how this works refer to Figure 8 which shows the noise/headroom performance of a typical equalizer with -90 dBu output noise and maximum output of +20 dBu. In Figure 8A , the equalizer is being driven by a signal level of +4 dBu (Ref. 0 dBu = 0.775 V). The signal-to-noise is 94 dB (90 + 4=94) and the headroom is 16 dB (+20-(+4)=16). In Figure 8B, the same equalizer is being driven by a signal level of -10 dBu. The signal-to-noise has degraded by 14 dB to 80 dB (90+(10)=80) and the

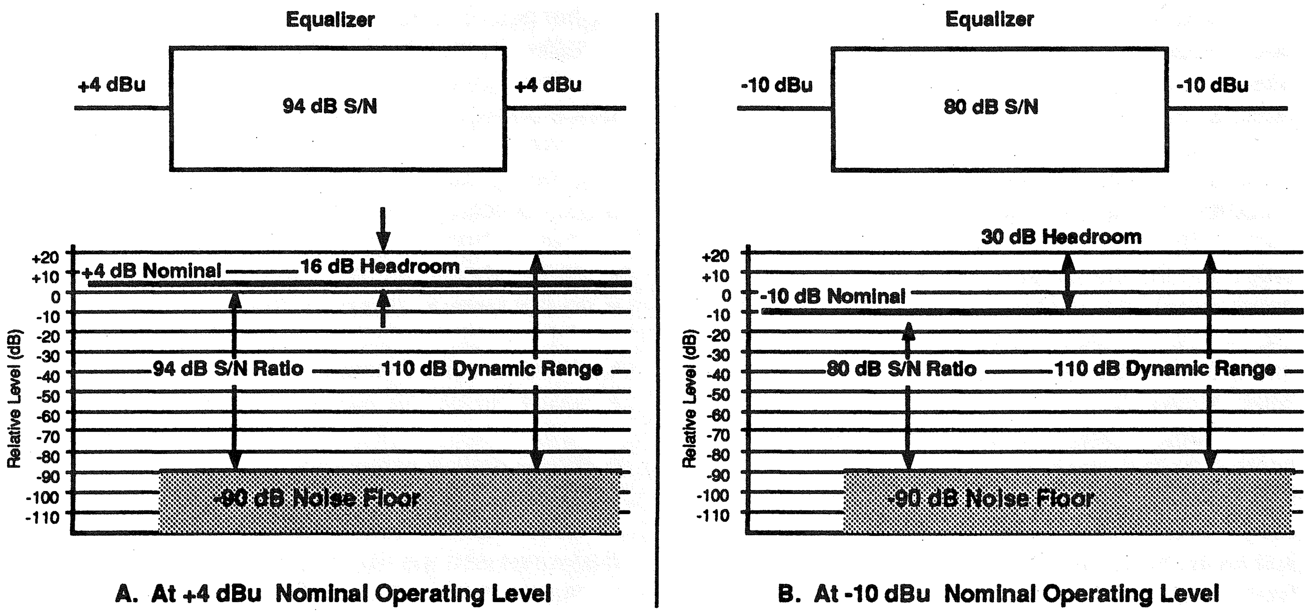


Figure 8. **Headroom and S/N Performance in a Typical Active Equalizer at Different Operating Levels**

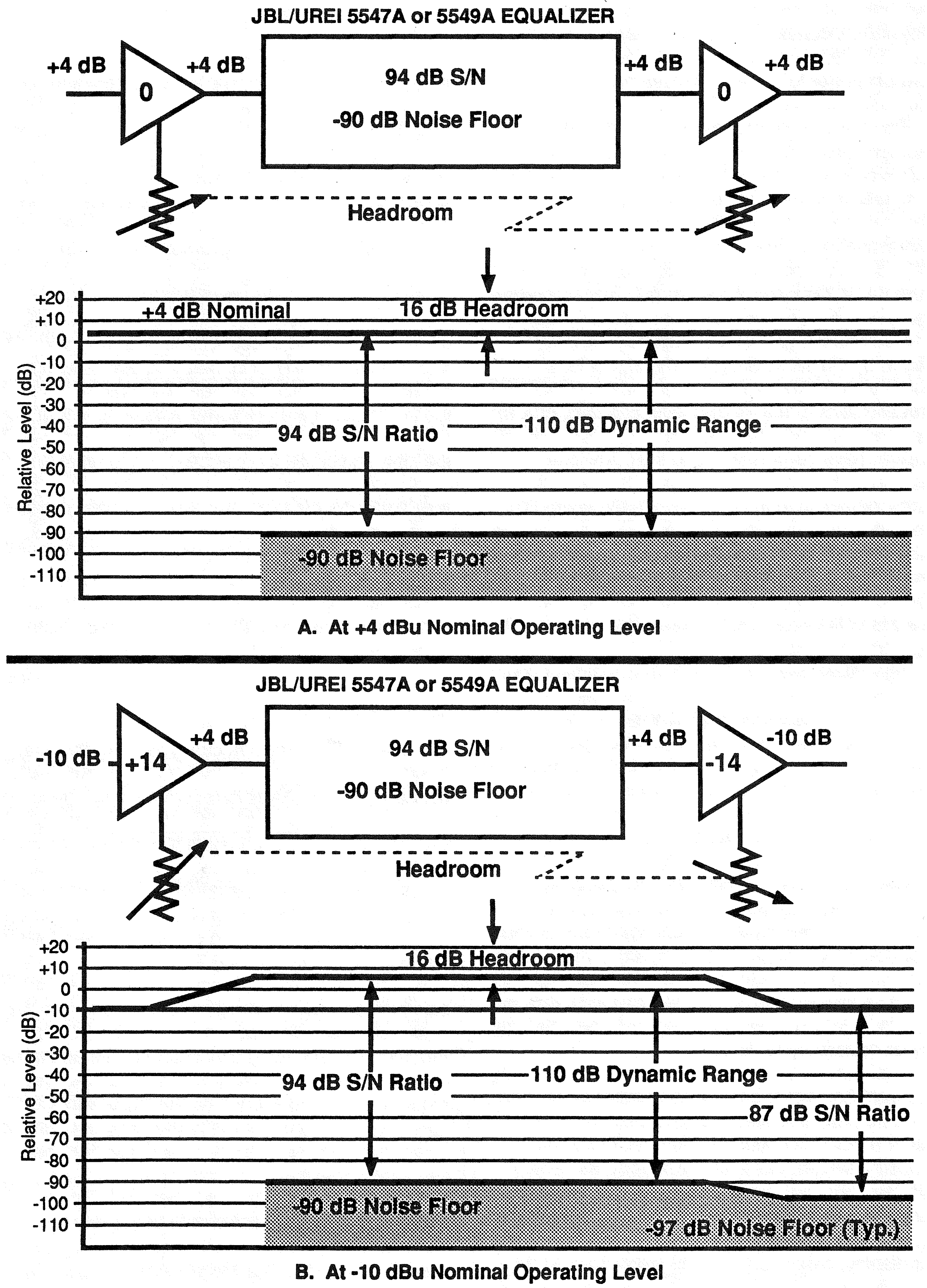


Figure 9. **Headroom and S/N Performance in a JBL/UREI 5547A or 5549A at Different Operating Levels**

headroom has increased by 14 dB to 30 dB (+20 -(-10) =30). Occasionally 30 dB of headroom is appropriate but in many situations it is excessive and we would prefer to trade off excess headroom for better noise performance.

Figure 9 shows what happens in the JBL/UREI headroom circuit for different levels. Note that a variable gain amplifier is inserted before and after the equalizer. In the case of a +4 dBu input signal the gains of the two amplifiers are set to unity and, as shown in Figure 9A, the noise and headroom numbers are unchanged from the example in Figure 8A. In the case of the -10 dBu signal however (Figure 9B), the gain of the input amplifier is raised by 14 dB and the gain of the output amplifier is reduced by the same amount. The signal level actually seen by the equalizer section has now increased back to the +4 dBu level and the signal to noise and headroom numbers at the output of the equalizer have been restored to their previous values as in Figure 9A. The output attenuator then returns the signal to the original -10 dBu level. The output attenuator improves the noise output from the device as a whole because it not only attenuates the signal from the equalizer section, but attenuates the noise output from the equalizer section as well. It does this until it reaches the noise floor of the output stage (about -97 dBu).

Now, of course, all of this gain adjustment could be performed external to the equalizer itself, but that would require additional amplifiers, pads and wiring with the attendant increase in circuit complexity and possible reduction in system reliability. It is much more convenient and cost-effective to include the facility right in the equalizer. Two adjacent front-panel linear slide pots serve as gain controls for the input and output stages with unity to +20 dB of gain and unity to -20 dB of attenuation, respectively. If the controls are moved together the gain through the equalizer does not apparently change to the outside world, but it can make a significant difference to the noise of the system. Of course the controls may be moved separately (that's why we provided two controls instead of one) and in some instances it may be necessary to do so.

Traditionally the adjustment of controls affecting headroom has either been a hit-or-miss proposition or one that required test equipment and time. Neither method is optimum. We have therefore included a simple, but effective method for setting the headroom controls. A peak reading LED display of available headroom is positioned next to the headroom adjustment controls on the front panel. This display, calibrated in 10 dB steps from 30 dB to 0 dB (clipping), gives the operator immediate visible indication of peak signal level and allows the adjustment of headroom in less time than it takes to explain or read.

End Cut Filters:

Although there are some who demand that the frequency response of their audio system should extend 'from DC to Channel Five,' restriction of frequency bandwidth on a selective basis is not only reasonable, but often necessary. The human body is not capable of sensing acoustic input over that wide a bandwidth, even if the system transducers were able to reproduce it. There are numerous situations where the restriction of bandwidth will improve the sound – not degrade it. For example acoustic leakage from one instrument in a studio to the microphone of another instrument. Also, restriction of the passband of a loudspeaker system may improve intelligibility or increase power handling capability by restricting out-of-band energy. As a matter of fact, there are some occasions where the only equalization needed is some judicious roll-off of the high and/or low frequencies. In the recording studio environment this means that the equalizer may serve double duty. The end cut filters of these equalizers are 12 dB/octave and are tunable through a wide range. In addition the High Cut filter (low-pass) may be switched to 6 dB/octave to allow more gentle tailoring of a House Curve. This last innovation was pioneered by UREI and has since been copied by others because of its utility.

Connectors:

Three different connector types give the user flexibility in wiring of signal input and output. Each connector has features to recommend its use in different situations. For fixed installations, we believe that the connector of choice is the barrier strip for several reasons. It is a known reliable connector which needs only a screwdriver and a pair of wire strippers to use. It does not require purchase of any mating connector, although crimp or soldered lugs or fanning strips may be used if desired. The connection is high-pressure, gas tight, and not subject to oxidation or accidental disconnection. It is also the least costly for the installer to deal with, both in terms of cost of mating connector (if any) and labor cost to install. On the other hand, the screwdriver is required each and every time the signal wires are connected or disconnected unlike the other connectors, which require no tools once the system has been initially wired.

Many people prefer the three-pin XL-style connector for its convenience in being able to connect quickly from one piece of equipment to another, especially in a system which is re-configured on a regular basis. The XL style also allows rapid hard wire bypass of a defective (or suspected defective) unit in a system which has no patch bays. On the minus side, the connectors cost several dollars apiece and require some time and effort to wire with a soldering iron. Also they are not recommended for long term use in situations where they are not plugged and unplugged occasionally as they can oxidize and give problems for low signal voltages, especially in areas of high RFI (Radio Frequency Interference).

The1/4 inch phone plug and jack have found wide acceptance in the music industry and the home recording market. The connector is widely available, relatively inexpensive and is used to connect a wide range of equipment types. Connecting cables use a plug on both ends with jacks being installed on the equipment relieving the installer of the necessity of checking connector compatability as on the XL type. An additional advantage is the capability of using a two conductor phone plug in a three conductor jack to automatically unbalance an otherwise balanced line. Phone plugs have the disadvantage that, unlike the XL style, the input and output cables of a piece of equipment may not be removed and plugged directly together to bypass a defective unit without the use of an adaptor. In elaborate systems, where cable shields need to be tied off at specific ends of cables for ground loop prevention, the interchangeability of ends of phone jack cables may prove a detriment rather than an advantage.

Bypass:

The Models 5547A and 5549A incorporate two different methods of circuit bypass for the two different purposes of bypass: circuit or AC mains power failure, and EQ in-out audition. Bypass on circuit failure ensures that program material will continue uninterrupted even if the equalizer fails during the program. This is accomplished via a relay with bifurcated gold contacts which connects the output terminals directly to the input terminals on failure of the AC supply voltage. If a failure occurs inside the equalizer but without an AC supply failure (for example: an op-amp going bad with resultant noise output) the power switch may be turned off and the signal will bypass the equalizer. The relay circuit has a short turn-on delay to prevent any possible power-on transients generated within the equalizer from being coupled to the succeeding equipment.

The second type of bypass incorporated is for auditioning the effect of equalization. This circuit, activated by a front panel pushbutton, bypasses only the filter sections while retaining input and output ampifier buffers and gain control. The buffers are kept in circuit so that the interface to preceeding and succeeding equipment is retained with no chance for gain shift or ground loop to occur as EQ is switched in and out. Some other equalizers provide only a hard wire bypass of the entire unit for this purpose which requires that the wiring be made compatible for both modes. This may not always be convenient to do. The EQ Bypass circuit allows the effect of any change in equalization to be easily heard without distracting level changes.

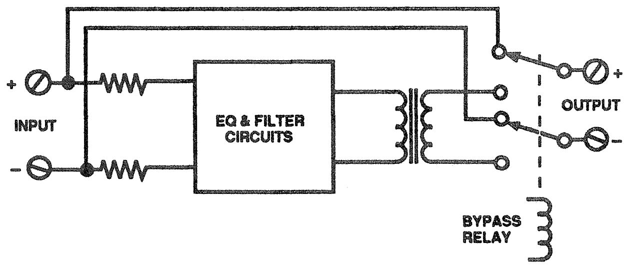


Figure 10. **5547A/5549A Relay Bypass Circuit**

There is actually a third bypass circuit just for the End Cut Filters. The End Cut Filters may be completely removed from the signal path so that they have absolutely no effect on program material. Many equalizers only allow the filters to be tuned to the widest bandwidth position to 'remove' the filters from the circuit. While this allows most of the amplitude variation to be eliminated, some phase shift will still occur at the extremes of the passband.

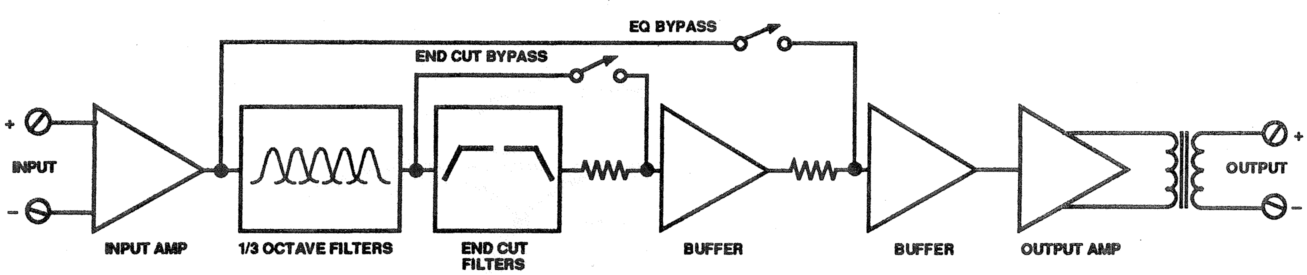


Figure 11. **Bypass Circuits for Equalization and End Cut Filters**

Conclusion:

We have discussed several of the circuit and component design problems peculiar to the one-third octave graphic equalizer. We have also discussed the reasoning behind several of the design decisions made during the development of the JBL/UREI Models 5547A Graphic and 5549A Room Equalizers. The design goal was to achieve a level of performance which was truly comparable to the best of the graphics designed with wire-wound inductors, but at a substantially lower price. We have met that goal.

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