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PROFESSIONAL AUDIO CONTROLS

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ALTEC LANSING

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Professional Audio Controls

The authors have long been recognized as outstanding authorities in the audio field and are members of the Altec Lansing staff.

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THE PROFESSIONAL AUDIO ENGINEER has long used fixed-gain/passive-control components in recording, broadcasting, and reinforcement system design.

Increasingly, the audio experimenter and high-quality-audio buff is becoming aware of the differences between home high fidelity components and those used by the professional engineer.

Three major differences are obvious when professional audio systems are compared with the very best home hi-fi systems. Professional systems utilize:

1. Fixed gain amplifiers.
2. Passive control devices.
3. Low impedance transmission circuits.

Not so immediately apparent are the factors of greater precision, longer service life, and greater versatility of service inherent in the professional components.

There are two types of controls possible in audio systems:

1. Active circuits.
2. Passive circuits.

Active Circuits

Active circuits are those that exert the required control by means of varying changes in the feedback circuit of an amplifier. Usually such controls are basically some form of frequency-discriminating network in the feedback loop of a negative-feedback amplifier which shapes the gain-frequency characteristics of the total amplifier.

Passive Circuits

This article is a short general discussion of the passive controls used in professional audio systems and shows how the advanced high fidelity buff can graduate to a fixed-gain, passive-control, professional sound system.

Passive controls are those that exert the desired change without power being added in the process. Passive devices take the form of:

1. *Attenuators.* Normally a resistive network that attenuates equally at all frequencies within the bandpass of interest while maintaining constant terminal impedances.

2. *Equalizers.* Devices that attenuate or restore inserted attenuation at selected frequencies.

3. *Filters.* Filters and special effects devices under this category are high- and low-pass filters and notch filters. These devices are characterized by zero insertion loss at all frequencies other than their operating frequencies.

Passive Equalizers and Filters

Passive equalizers and filters can be of the RC, LC, or RLC type. They can be constant "K," "M" derived, Butterworth, or Chebishev design. The circuits employed can be:

1. Series 2. Shunt	Both of these circuits are characterized by input and output impedances that change with frequency.
3. Full series 4. Full shunt	These circuits maintain a constant input impedance but the output impedance varies with frequency.
5. "T" 6. Bridged "T" 7. Lattice type	These three circuits are identified by constant input and output impedances.

Equalizers and filters can further be of the balanced or unbalanced circuit configuration. Because of the 2-to-1 cost consideration and the very short length transmission paths inside a professional audio console, most equalizers and filters are of the unbalanced type. Any transmission circuits that enter or leave the console, however, are balanced and isolated from the unbalanced console circuitry by the use of high-quality repeat coils (isolation transformers).

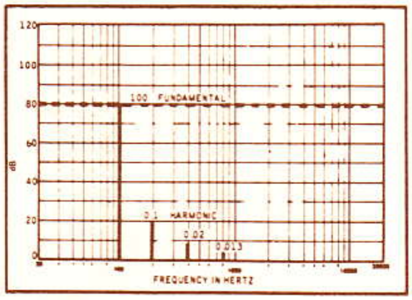


Fig. 1. True distortion of a circuit with the frequency response of Fig. 2.

Circuits Used Commercially

The devices discussed in this article are all of the four-terminal constant-K, unbalanced-circuit, passive, bridged-T type. The preferred impedance is 600 ohms, though 150 ohms as well as other values are encountered in the use of such devices.

Advantages of Passive Controls

The advantages of using such a passive control can be summarized as follows:

1. Very low design cost compared to a really stable active circuit.
2. Less complicated to build and to use.
3. More reliable over long time intervals. Greater repeatable accuracy.
4. More versatile. It can be used in system after system because it is an independent component.
5. Extremely low maintenance costs.
6. Minimum distortion that remains constant. No circuit changes in amplifiers will change the inherently low distortion figure of the controls. (Not measurable.)

Precautions to Observe

It is possible in the design of active circuits to "magnify" the "Q" of an inductance used, thereby allowing smaller sized components to be employed. When passive devices are used, particularly at low frequencies, size and weight of the components required will not lend themselves to miniaturization. Due to the high "Q" coils employed in passive equalizers care should be exercised when they are employed near appreciably high flux fields.

Distortion Measurements Pose Special Problem

Finally, it should be recognized that any frequency-response curve with a changing amplitude characteristic makes distortion measurements difficult. A typical example is the measurement of distortion in a crossover-network where the technician begins at a frequency that is on the slope of the network's response. Let's say that the real distortion characteristic looks like the chart shown in *Fig.* 1. *Figure* 2 discloses the slope of a 12-dB-per-octave crossover network. If 100 Hz is chosen as the measuring frequency, then it can be seen that the second harmonic will be increased 12 dB due to the slope. From *Fig.* 3 it is obvious that a harmonic 60 dB below the fundamental represents 0.1 per cent distortion, but, due to the slope of the network, the 2nd harmonic is now 60 dB minus 12 dB below the fundamental or 48 dB below the fundamental which equals 0.4 per cent distortion. The 3rd harmonic was 74 dB below the fundamental or 0.02 per cent distortion. Now, because of the slope of the network's frequency response it is now 74 dB minus 24.5 dB below the fundamental, or 49.5 dB below the fundamental, and this equals a distortion indication of 0.32 per cent.

Measuring Distortion in Equalized Circuits

This same discrepancy can occur in the use of high frequency equalization. *Figure* 4 illustrates the distortion characteristics of a system at a "flat" setting, while *Fig.* 5 shows the apparent increase in distortion caused

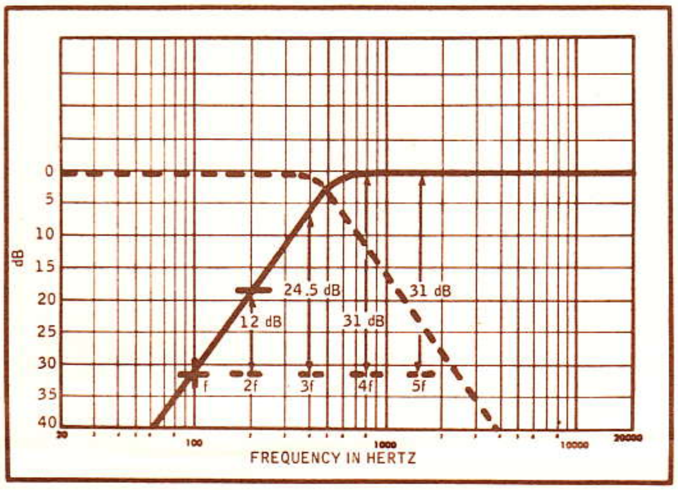


Fig. 2. Effect of frequency response on measured distortion figures.

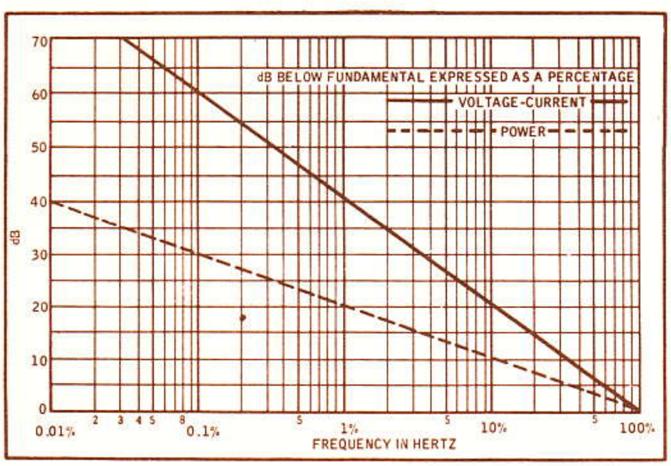


Fig. 3. Chart showing percentage derived from measurement of distortion as the number of dB below the fundamental.

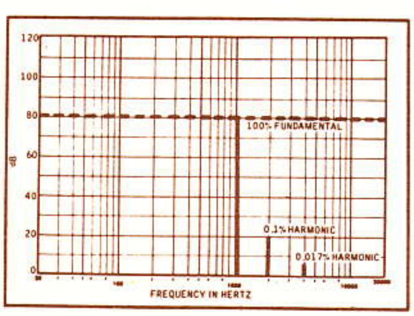


Fig. 4. Distribution of harmonics in a flat-response system.

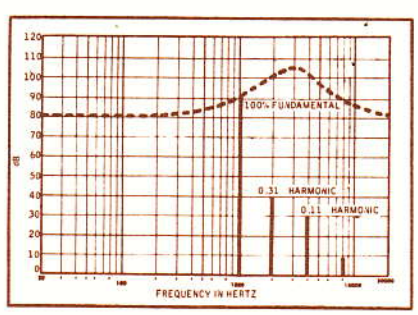


Fig. 5. Distribution of harmonics in a system with a "boosted" response.

by a high frequency boost of 12 dB at 3000 Hz. (Notice that the 100 per cent indication for the fundamental is also raised by the equalization process.) So far as the ear is concerned such boost sounds like a real increase in distortion and this effect puts a practical limit on high-frequency boost of about 12 dB.

Low-frequency boost can have the inverse effect by dropping the higher harmonics still lower in relation to the fundamental .(This may well be the reason low-quality package hi fi's are usually adjusted "bass" heavy by their users.)

Calculation of Correction Factor

To obtain an accurate harmonic distortion figure from a system with a rising high-frequency amplitude characteristic, take the measured distortion expressed in dB below the fundamental and add the dB increase due to the slope of the system. This total, converted to percentage, will equal the true distortion of the system. (dB measured distortion + dB slope increase = dB below fundamental.)

Other Considerations

Attention must be paid to the insertion loss of the device (equalizer) and the maximum and minimum levels required for satisfactory operation. In passive equalizers there are two choices that can be made regarding insertion loss. Because it is a passive device the only way to obtain "boost" is to attenuate all but the desired "boost" frequency. When this is done the "boost" frequency is higher, relatively, than the non-"boosted" frequencies. However, if we merely depressed the frequencies to be attenuated in order to "boost" the selected frequency, we would have a loss in level at all frequencies attenuated that would vary with each equalizer setting.

One way to achieve constant gain in the equalizer is to use an inverse loss attenuator in conjunction with the equalizer. The attenuator inserts loss at the "flat" setting and gradually removes it to the same degree that the majority of the bandpass is depressed to obtain equalization.

Preferred Method

In the equalizers to be discussed in this article a full bandpass insertion loss is chosen, usually 14 to 16 dB, and equalization restores the required amount of this loss to obtain selective boost at chosen frequencies. By this means the only level change encountered in the program material is that occasioned by "boosting" a selected frequency. The resulting increased level at that frequency is the sought-after effect. The unequalized portion of the bandpass remains at a constant level.

Dynamic Range Determined

Passive equalizers and filters of the unbalanced, bridged-T, constant-K type illustrated in this article should be operated between the levels of —70 and +20 dBm. *Figure* 6 reveals the dynamic range, noise, and gain characteristics of a high-quality, fixed- gain, plug-in-type professional audio system amplifier. This same amplifier shown in *Fig.* 7 can be used as a preamplifier, booster, program, and even line amplifier by merely changing the "strapping" at its *external* socket connection. This allows a single type plug-in unit to be used interchangably anywhere in the system and this minimizes the number of units required for maintenance back-up. Fixed-gain amplifiers of this type can be designed for optimum negative feedback and stability. *Figure* 6 shows that the maximum output for the unit (at less than 1 per cent THD 20 to 20,000 Hz) is —27 dBm. Since it has a *fixed voltage gain* of 45 dB terminated and 51 dB unterminated, the maximum allowable input signal must not exceed —18 dB and —24 dB respectively.

With these figures in mind it can be seen that the maximum input signal into an equalizer placed just ahead of such an amplifier must not exceed —16 dB. (—16 dB)—(14 dB insertion loss) + (12 dB max. equalization) = —18 dB output at some selected frequency. If for one of a number of reasons the output from the equalizer exceeds this figure the use of a fixed loss pad can correct the level.

Fixed Parameters

Again referring to *Fig.* 6 the fixed parameters are seen to be maximum output level, maximum input signal level and the equivalent-input-noise figure (E.I.N.), and consequently, the maximum signal-to-noise ratio. The dynamic range and the minimum signal-to-noise ratio are parameters that are played against each other depending on the requirements at hand. In general the minimum signal-to-noise ratio need not exceed 10 dB with 25 to 30 exceptionally quiet. The maximum dynamic range available in a

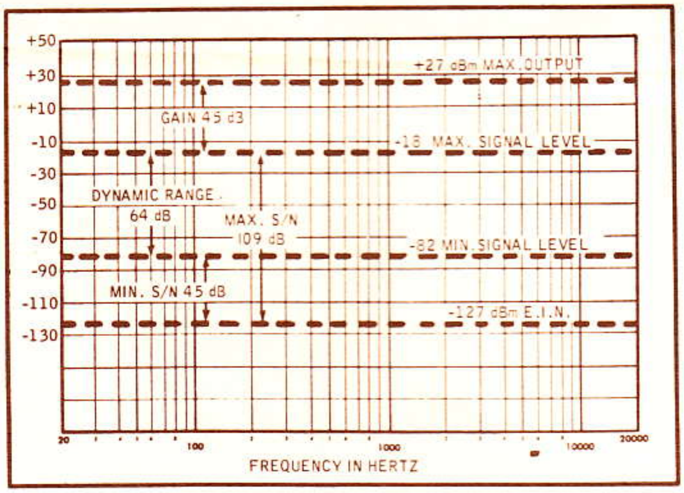


Fig. 6. Relation of dynamic - range, noise, and gain characteristics of a high - quality fixed-gain professional amplifier.

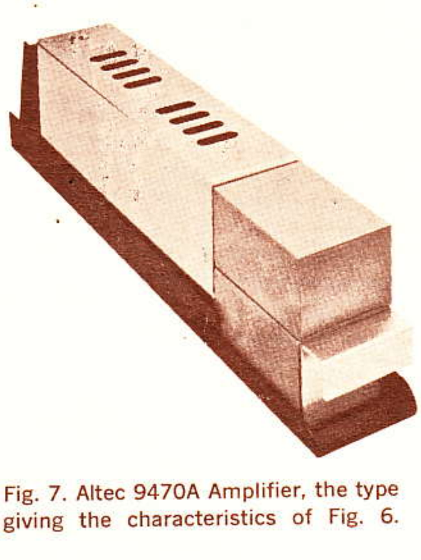


Fig. 7. Altec 9470A Amplifier, the type giving the characteristics of Fig. 6.

very quiet studio is shown in *Fig.* 8. The noise level shown is Noise Criterion Curve NC-20, and the maximum level shown in the maximum sound pressure level (SPL) in dB at 4 feet that can be expected from the best monitor speaker available. Since the figure shown, 93 dB, is the maximum dynamic range, it is also the maximum signal-to-noise ratio. Because the ear can barely hear sound below NC-20 any sound system with a maximum signal-to-noise ratio of better than 90 dB is going to have a noise level below that found in even the quietest acoustical environments.

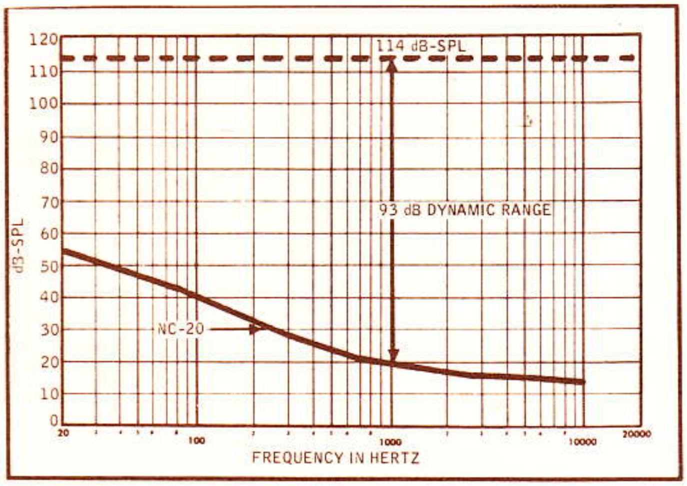
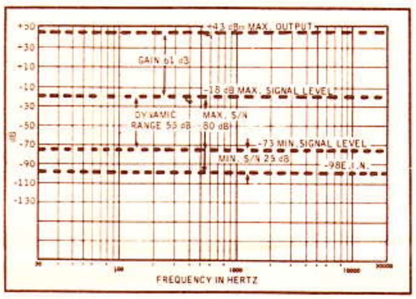


Fig. 8. Maximum dynamic range available in a quiet studio. The noise level shown in the Noise Criterion Curve NC-20.



Fg. 9. Characteristics of a typical 20-watt professional-type monitor amplifier Altec 9471A, shown in Fig. 10.

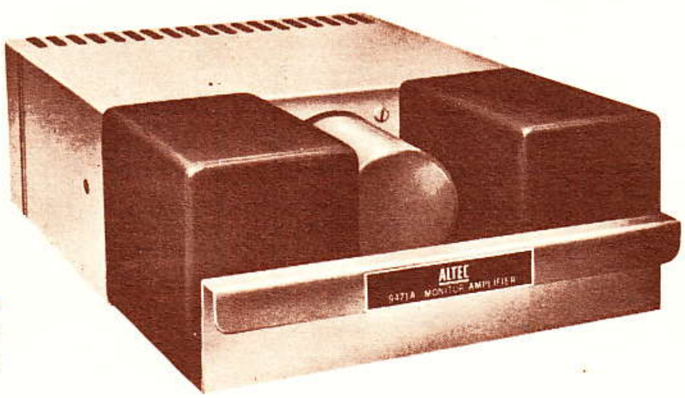


Fig. 10. The Altec 9471A, 20 - watt monitor amplifier.

The component that usually determines the noise threshold of the electronic part of the system is the power amplifier. *Figure* 9 gives the gain, maximum signal level, and maximum signal-to-noise ratios for the 20-watt professional monitor amplifier shown in Fig. 10.

Practical Limits to Dynamic Range

Figure 11 shows the dynamic range, and maximum and minimum signal-to-noise ratios required by a major recording company for their recording consoles, master tape recorders, master disc lathe, and finally, their production pressings. In the final analysis, the real determination of realistic dynamic range requirements and signal-to-noise specifications is the limits set by the source material. Even the finest professional tape recorder has a maximum signal-to-noise ratio that is 10 dB less than that achieved in the 20-watt amplifier shown in *Fig.* 10.

Another limitation of dynamic range in a listening system is the occasional use of a small "bookshelf" type speaker system with a mass loaded cone to help lower cone resonance in a too-small enclosure. Such loudspeakers exhibit a minimum level below which the inertia of the cone and the resistance of the suspension are not overcome by the signal. Once the input signal reaches a level of sufficient magnitude to overcome the friction and inertia, the loudspeaker suddenly "comes on." The transition from low-level output to no output is not continuous but is characterized by the loudspeaker "shutting off" below a critical level. The solution, of course, is replacement by a more efficient type of loudspeaker utilizing a lighter moving system.

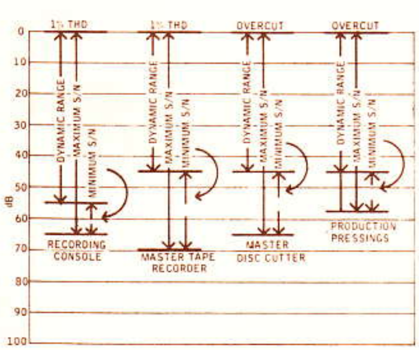


Fig. 11. Characteristics specified by a major recording company for various elements of their recording channels.

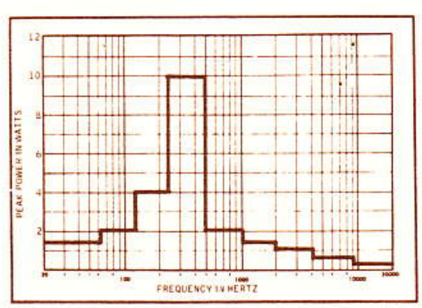


Fig. 12. Peak power per octave in a typical orchestral composition.

Energy Distribution of "Live" Source

One final consideration of dynamic range and equalization is that of the distribution of energy in typical source material. *Figure* 12 exhibits the peak power expressed in watts *vs.* frequency of a typical orchestral composition. It can easily be seen that consideration of equalization in the region of 125 Hz to 500 Hz requires careful consideration of maximum allowable input signal to the amplifier that follows such equalization. (For best results each program source should be individually studied to determine frequency distribution of energy.) Æ

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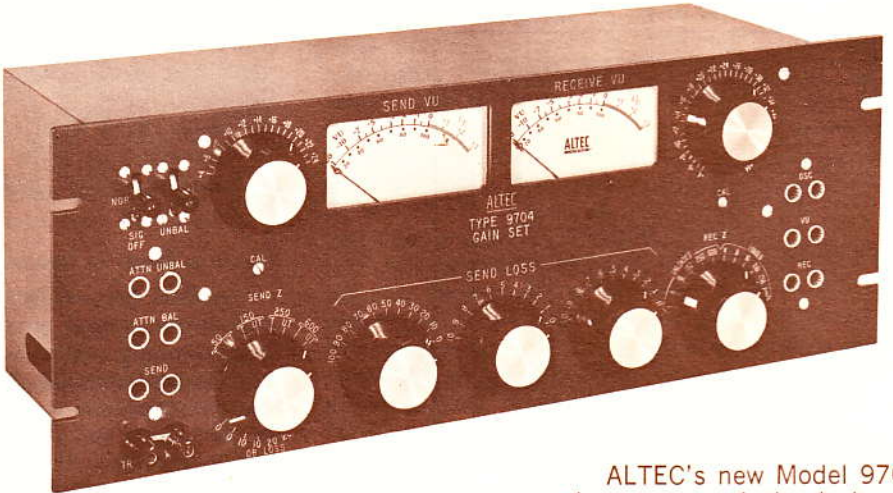

ALTEC's new Model 9704A transmission measuring set, precisely designed to accurately measure gain, loss, frequency response and signal levels of individual audio devices or complete installations.

Typical Equalizer and Filter Circuits

THE BASIC EQUALIZERS AND FILTERS, and the constant-K, unbalanced, bridged-T circuit for them, along with a sample response curve, are shown in *Figs.* 13 through 22.

A high-frequency rolloff is shown in *Fig.* 13, and a low-frequency rolloff is shown in *Fig.* 14. These are the types of circuit that is used in program-equalizer attenuation settings.

Figures 15 and 16 illustrate the circuits used to produce special effects or to eliminate interference that has a definite spectral frequency. The broadness or sharpness of these curves can be varied by a change in the L-C ratios.

Figures 17 and 18 give the circuits that are used to restore deliberate insertion losses to "shelved" type accentuation.

Figures 19 and 20 show frequency-selective attenuation and "boost" circuits. It is this type of circuit that is used in a typical high-frequency-boost-or-cut portion of a program equalizer or graphic equalizer.

Figures 21 and 22 consist of an 18-dB-per-octave low-pass and high-pass filter. These are extremely useful circuits for limiting the passband of a wide-range system to those frequencies which the transducers involved are capable of handling cleanly, thereby avoiding overload from frequencies not containing useful energy. Variable equalizers and filters are merely series of these basic circuits sequentially selected by means of precision switches such as shown in *Fig.* 23.

Equalizers and filters of this type require careful termination in their rated impedance. *Figure* 24 diagrams the test set-up used to record the frequency response curves that follow.

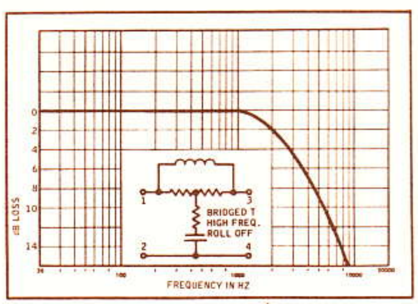

Fig. 13. Bridged-T high-frequency roll-off equalizer and its response curve.

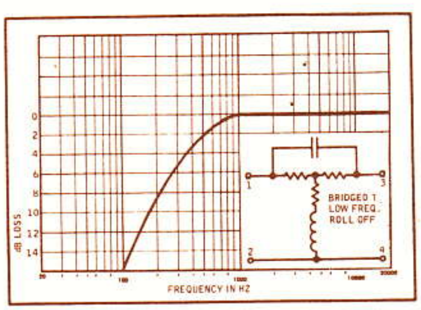

Fig. 14. Bridged-T low-frequency roll-off equalizer and its response curve.

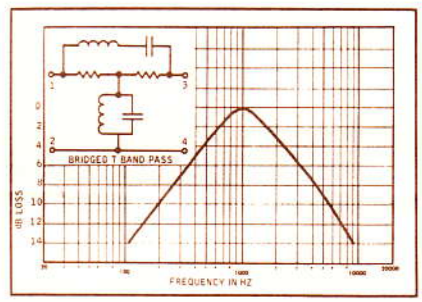

Fig. 15. Bridged-T band-pass equalizer and response curve.

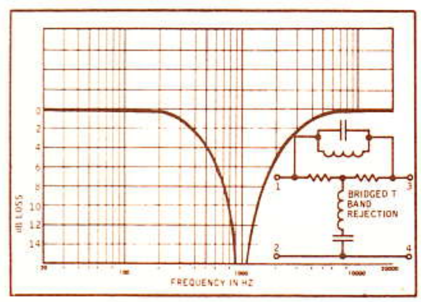

Fig. 16. Bridged-T band-rejection filter and response curve.

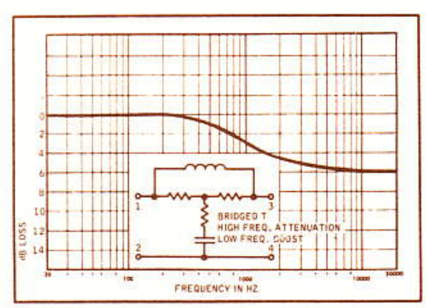

Fig. 17. Bridged-T "shelving" equalizer used to attenuate highs, boost lows, and the resulting response curve.

Design Laboratory Illustrated

The use of a servo-operated recorder-generator assembly allows automatic recording of the amplitude-frequency characteristics of equalizers and filters. Due to the reduced time such testing requires as compared to conventional hand plotting (and the reasonable cost of such test instruments today), high-quality professional equalizers and filters now come with such curves recorded for each individual unit. *Figure* 25 shows a laboratory where equalizers and filters are designed and measured. The output of the recorder-driven test oscillator shown at the lower right corner is flat within ±0.5 dB over its entire frequency range.

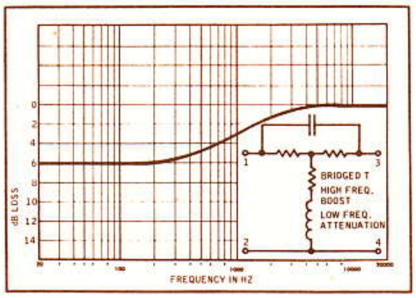

Fig. 18. Bridged-T "shelving" equalizer used to boost highs, attenuate lows, and its response curve.

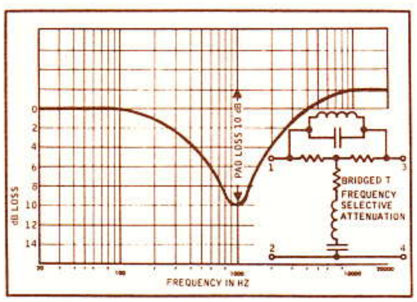

Fig. 19. Bridged-T frequency-selective equalizer used to eliminate or reduce unwanted peaks in response or to produce a desired effect.

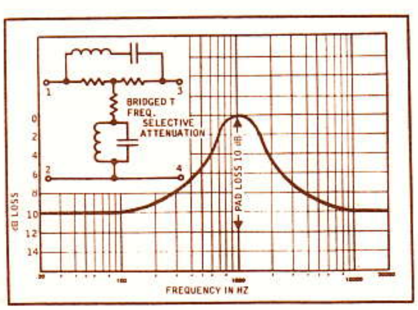

Fig. 20. Bridged-T frequency-selective equalizer used to increase output at a specific frequency.

Commercially Available Passive Controls

Commercially available versions of passive equalizers and filters are available from a number of manufacturers. They take the form of microphone equalizers, program equalizers, graphic equalizers, filter sets, and fixed pre- and post-equalizers.

Microphone or Dialogue Equalizer

A microphone equalizer and its plug-in mounting bracket are shown in *Fig.* 26. This is a series of passive LCR bridged-T, constant-K networks with input and output impedances of 600 ohms. The insertion loss at the "0" or flat setting of the controls is 14 dB. It provides low-frequency shelving at 100 Hz and high-frequency boost at 7 kHz. Maximum boost available at each of these frequencies is 12 dB. It also provides selective attenuation in 2-dB steps up to a maximum of 16 dB at 10 kHz and 16 dB at 100 Hz, as shown in the curve of *Fig.* 27.

This figure shows the frequency-response curve of it at its flat setting, and then one run at maximum bass boost/maximum treble roll off, and another at maximum bass roll off/maximum treble boost.

Program Equalizers

Figure 28 shows a more sophisticated version of the microphone equalizer which is known as a program equalizer. The internal construction showing the mechanical means

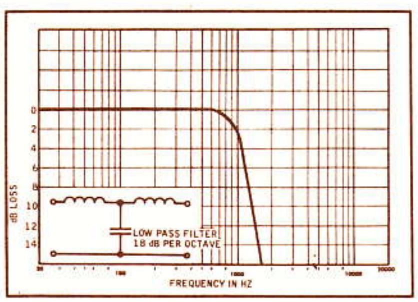

Fig. 21. 18-dB/octave low-pass filter and its response curve.

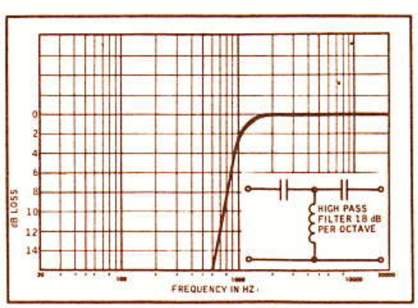

Fig. 22. 18-db/octave high-pass filter.

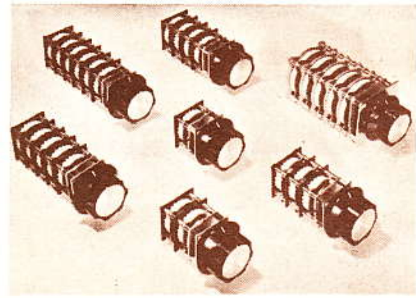

Fig. 23. Precision switches used for attenuators and equalizers in professional audio systems.

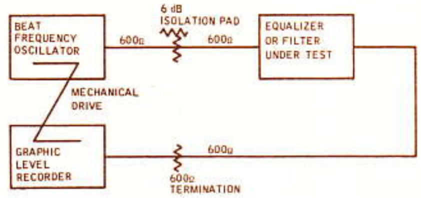

Fig. 24. Block diagram of measurement method employed to record response curves automatically.

of selecting the large number of network-circuit combinations represented illustrates why most of those in use are commercially manufactured units. Equalizers like the microphone unit previously shown are used principally as dialogue equalizers or to correct room problems in a sound stage or studio. Program equalizers, on the other hand, are designed with the playback system in mind as well as the recording system. In addition to its attenuation and boost functions, in 2-dB increments, it allows a choice of two low-frequency shelves at 100 and 40 Hz at the calibrated points and four high-frequency-boost points—3, 5, 10, and 15 kHz. The functions desired are selected by the two knobs at the top of the panel.

On the low-frequency shelving switch are two "off" positions that bypass the entire equalizer and insert a 14-dB loss pad in its place. This allows instant aural comparison between equalized response and "flat" response at the turn of a switch.

Amplitude Response Curves

Figure 29 displays a composite frequency-response chart of the low-frequency shelving at 40 Hz and high-frequency boost at 10 kHz for different amounts of attenuation at each of the steps available. Note that the shelving control affects only the boost portion of the equalizer's response. Similar curves would result from the

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100-Hz setting, but would simply be moved up the frequency scale about 1½ octaves. The high-frequency boost selector was set at 10 kHz although 3-, 5-, and 15 kHz positions are also available. Note here also that the high-frequency boost selector affects only the boost functions of the equalizer. These units are usable in any 600-ohm circuit by simply inserting them in the transmission line at the proper level. A wide variety of response curves can be obtained with this type of equalizer.

Graphic Equalizers

A graphic equalizer is shown in *Fig.* 30. This equalizer allows exceptional control of a signal from 50 to 12,500 Hz. The unit shown consists of 7 boost and 7 attenuate equalizers covering a range from +8 and −8 dB. The total unit has an insertion loss of 16 dB with all controls set at "0" or "flat" position. Each control operates in 1-dB increments. Again, the circuits employed are bridged-T, constant-K, passive networks, designed for use in 600-ohm lines. The special effects possible with this unit range from an "other worldly" voice of the ghost in Hamlet to increased "presence" on a distant microphone pick-up. The operator's imagination is about the only limitation the instrument imposes. The curves of *Fig.* 31 were recorded by measuring each boost setting and each attenuate setting individually at its top and bottom position. This enables one to see the individual filter shapes associated with each control. Where the filter "skirts" cross, interaction occurs if two adjacent controls are used simultaneously. This allows smooth continuous curves to be formed.

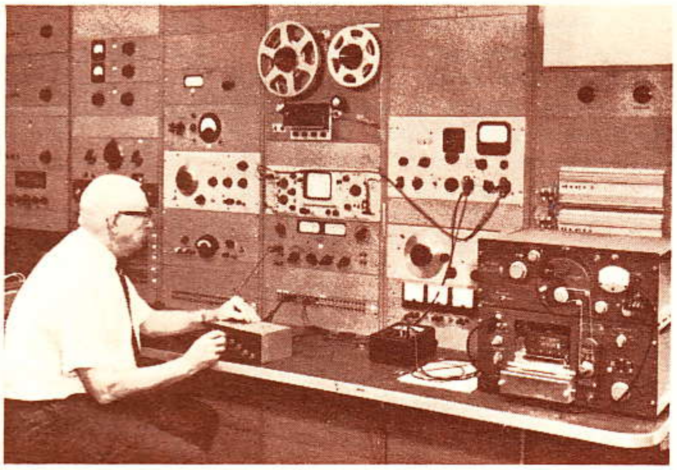


Fig. 25. Co-author Arthur Davis measures response of a graphic equalizer. Note automatic curve-tracing equipment at lower right corner.

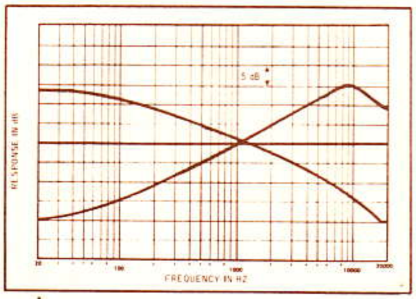


Fig. 27. Limits of curves obtainable with the microphone equalizer of Fig. 26.

Fig. 30. Graphic Equalizer which will permit boost or cut of 8 dB at any of seven frequencies approximately $1\frac{1}{2}$ octaves apart. One, two, three, or all of the controls may be varied to achieve the desired effect.

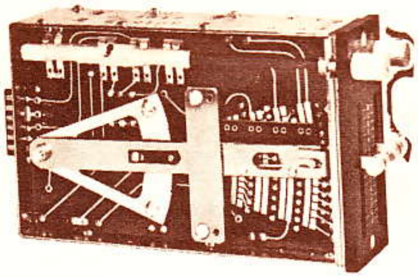


Fig. 28. Program equalizer which will provide the wide variety of curves shown in Fig. 29. Any combination of these curves may be obtained.

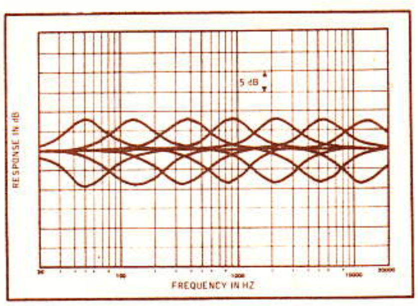


Fig. 31. Composite of 14 individual curves, each of which was run with a single control at its extremes. Superimposed, they appear as shown. If all controls were put at +8, for example, the resultant curve would be practically flat at the +8 level, and similarly would be practically flat at the −8 level if all controls were placed at −8.

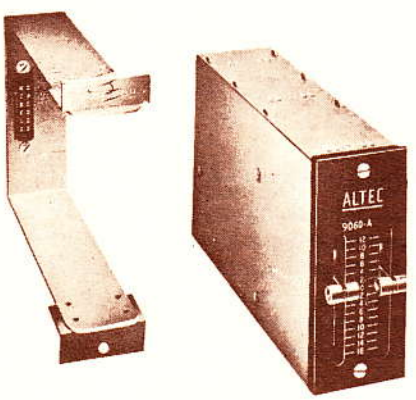


Fig. 26. Altec 9060-A microphone equalizer and its mounting fixture.

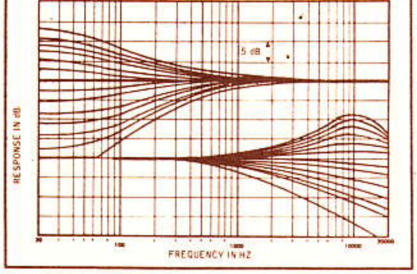
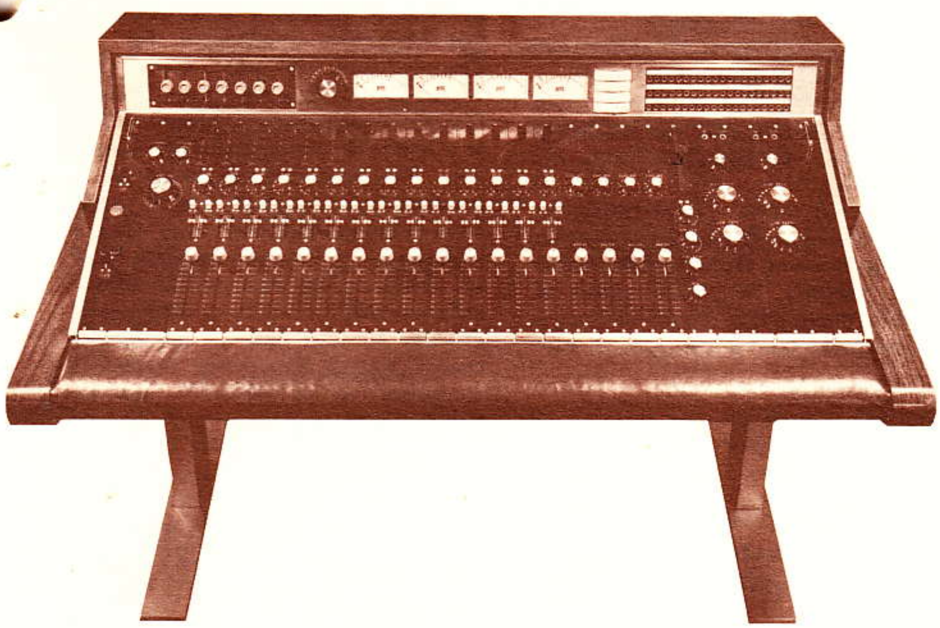


Fig. 29. Composite of curves obtainable with equalizer of Fig. 28.



ALTEC's sensational new Model 9200 professional console — a basic modular type enclosure for assembly by the user from Altec catalog components to meet his individual requirements. User purchases the basic cabinet and hardware, drilled, punched and beautifully finished — all difficult metal work is done. He then selects desired Altec modules to complete the console in accordance with his needs.

Fig. 32. Typical filter set providing eight cutoff frequencies for the high-pass section, and eight for the low-pass section. It is capable of producing response curves like those shown in Fig. 33.

Use of Filters

Once equalization is accomplished, consideration should then be given to the filters desired to maintain the integrity of the passband of interest to the user. A tremendous source of distortion and overload in audio systems lies in building the system to *try* to reproduce frequencies of no use to the program material actually desired. (This is not to be confused with the concept of designing very-wide-range components in the effort to minimize phase shift in the passband of interest, but refers to final total system response.)

In a quality system it is foolish to feed a well designed loudspeaker high-amplitude signals below its lowest rated frequency. When there is doubt about source material (turntable rumble cut into the record, eccentric record due to misplaced center hole, and so on) a high- and low-pass filter set, in this case, at the input, saves the system from amplifying, controlling, and attempting to reproduce gross defects.

Figure 32 shows such a filter set with multiple choices of cut-off frequencies. Such filters can also be constructed as fixed type units, but are obviously not as flexible in use.

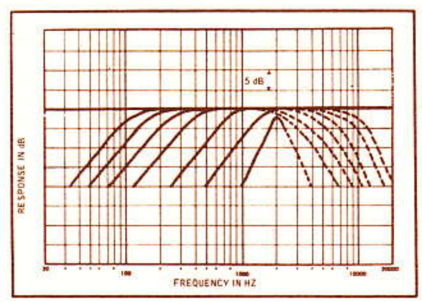


Fig. 33. Reponse curves available from the filter shown in Fig. 33. A wide variety of effects can be created with this type of filter.

Figure 33 shows the various cutoffs obtainable with different control settings. The high-pass curves are shown solid, the low-pass ones dashed. The overlap of the cut-off frequencies at each end of the spectrum allow the unit to be used as a simple form of bandpass analyzer. Here, once again, the circuit is a series of bridged-T, constant-K, passive 600-ohm networks wired to precision rotary switches in order to permit variable choices.

Filters are characterized by no insertion loss until their cut-off. F_c by definition is the point where the amplitude response has fallen 3-dB and these filters follow that convention.

Application of Passive Controls

If the components available seem numerous, it is because the possible applications in various types of systems are too many to tabulate. In order to discuss just a few possible applications to a practical system and show how the installation of such devices is handled, a small three-channel, passive-control, fixed-gain playback system with two inputs per channel was designed following accepted professional techniques.

Figure 34 is a single-line block diagram of the system. In working with such a system "audio levels" are carefully tabulated throughout each stage of the system, and plotted as a "gain chart." The one in *Fig.* 35 represents our simple system and is typical. Such a chart constitutes a gain and loss "road map" for the system.

This particular system was designed to accept either a tape-head input or a tuner input. The tape-head input feeds a fixed-gain preamplifier which is operated unterminated from the tape head.

Figure 36 shows several typical input situations involving fixed-gain amplifiers. About the only control that could properly be placed ahead of the first preamplifier would be a high- or low-pass filter if the source were suspect. Normally, however, the first passive control that will be encountered is on the output side of the first fixed-gain amplifier, as in the system of *Fig.* 34, where a fixed equalizer provides post equalization for the characteristics of the tape head (NARTB, CCIR, or the like). Following this equalizer is a booster which is a fixed-gain terminated-input amplifier. The booster is followed by a program-type equalizer (or it could be a graphic equalizer) and then a high-pass and low-pass variable filter set. Between the filter and the adjustable loss pad a bridging pad "bridges" the line to feed half of the derived center channel. The center channel is comprised of ½ (A+B); and its philosophical justification goes back to 1932 and the original Bell Telephone Laboratory Symposium on Auditory Perspective. (Those who would like to read the very convincing arguments for a derived third channel are referred to the bibliography at the end of this article.)

Following the loss pad is half of a differential rotary attenuator used as a balance control. A second booster is optional, but, if used, would be inserted just after the balance control. Note that the gain chart in *Fig.* 35 exhibits the effect with or without this booster. If the booster is not used, the adjustable loss pad is set at −7 dB instead of −36 dB.

The rotary attenuator that follows the booster is shown with 22 dB of loss. If the booster is not used, the gain is shown on the chart with the attenuator turned full on, or 6 dB of insertion loss. (This is a ladder attenuator.) If the amplifier has a bridging input it must be terminated with a 600-ohm resistor to insure proper tracking of the attenuator. The attenuator then feeds the power amplifier, and the power amplifier in turn drives the loudspeaker system. Power levels required for full output from the amplifier and the resulting sound power level (SPL), specified in dB four feet in front of the loudspeaker, are also calculated. The professional knows how loud his system will be at any setting long before he ever throws the first switch. (Ideally the gain of the system should be set so that normal attenuator settings are between −10 and −16 dB.) The center channel bridges each side channel with a 20,000-ohm bridging pad. These two signals pass through a mixing network (6-dB loss each leg but addition of the two signals brings out the mixed signal at the same level as each input signal to the network).

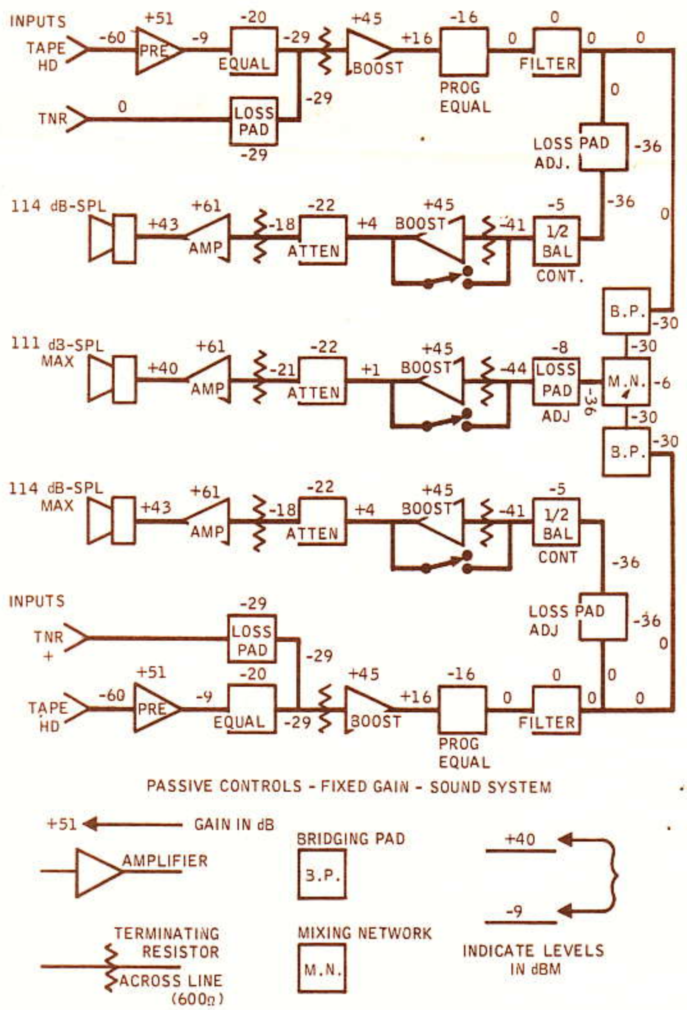


Fig. 34. Single-line block diagram of a typical system showing gains and losses of the various components, together with the signal levels at each connecting circuit.

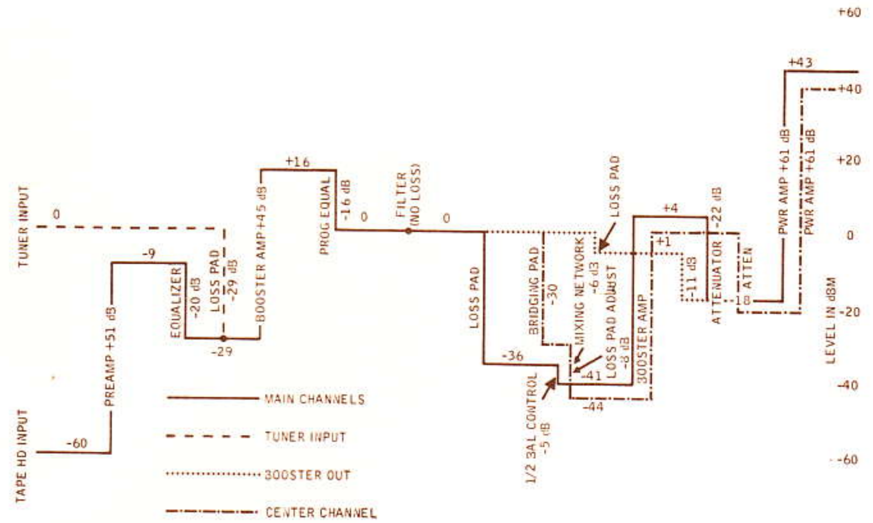


Fig. 35. Level diagram of the circuit of Fig. 34. This type of diagram gives an immediate picture of the various levels throughout the system—of greatest importance in the design.

At this point another adjustable loss pad is encountered which enables the center channel to be adjusted 3 dB lower than the two side channels. (Acoustical mixing of the two side-channel speakers provides a substantial acoustical middle-channel signal. If the physical middle-channel speaker is set to a level equal to the two side channels, its output plus the side channels gives an apparent output from the center that is too high.) The presence of adjustable loss pads in each channel allows for any gain balance desired or required under difficult room environments, or to allow different-channel gain balances to suit individual ears. Once the system is properly adjusted for a given space and use, the variable controls provide more than sufficient day-to-day and source-to-source adjustment.

Wiring & Grounding the System

Although this subject could easily fill a large book, a few suggestions may be helpful.

All wiring in audio systems of this type should be done with two-conductor shielded cable (usually with a solid shield wire in addition to the shield), and with the shield insulated. The low side of every network is carried through on one of the two conductors to the *single* system ground. All grounds are connected at the same spot. Shields are grounded at one end only and each run between components should be made with only *one* end of the shield being grounded. The other end of the shield is left *disconnected*. Even with these precautions, the wiring in such systems is relatively straightforward.

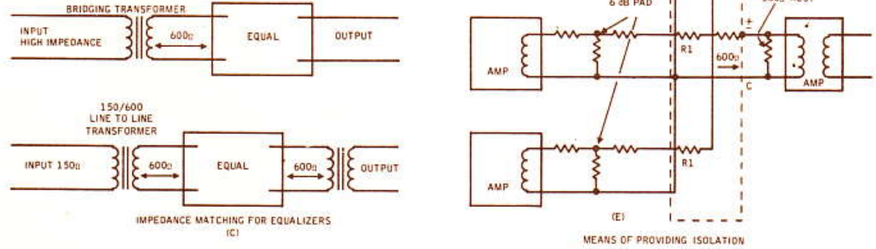


Fig. 36. Several acceptable ways of connecting passive equalizers into a system, and methods of providing isolation between circuit elements which might interact to cause response irregularities.

Summary

It is obvious that the professional audio engineer has a much wider choice of components than the home user. The components are also of higher quality and usually of far greater accuracy and re-setability. Best of all, as the years pass the old equipment isn't discarded. The basic quality components can be expanded or rearranged to meet new requirements.

It is hoped that the dedicated sound-system designer will consider stepping up from high fidelity components to the use of professional passive-control/fixed-gain playback systems with their advantages of better performance, longer life, and lower cost. Æ

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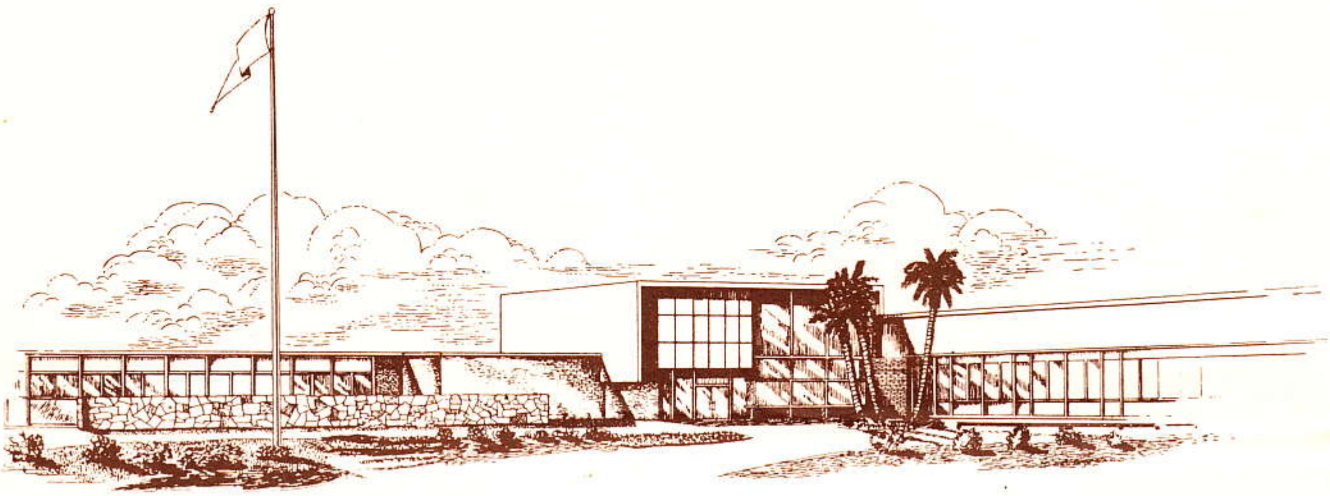
Reference to these works and the further bibliographies they contain will provide a good conceptual view of design requirements pertinent to passive attenuators, equalizers and filters.



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