SOUND

What Is Equalization?

A dictionary would tell you that to equalize is to make equal or uniform. Wally Parsons, ETI audio contributor, discusses.

ALL MEN (and women) are said to be created equal, but they don't all stay that way. Some become fat, others thin, some tall, some short, some rich, others poor; in other words, some are more equal than others. And the same is true with audio electrical and acoustical signals. Sure, they start off okay; but as soon as a musician pushes a sound out of his horn and sends it hurtling to a microphone little gremlins start chewing at it as it makes its tortuous way through the air, into the microphone, the bewildering maze of wires and transistors, cutters, pickups, loudspeakers, listening room. Indeed, it often seems miraculous that it emerges from all this as something even vaguely resembling the original. And then, of course, there is man, ever ready to show mother nature the errors of her ways, tinkering with this signal to make it conform to his own concept of perfection.

Then, too, sound was never meant to be recorded; therefore we have no choice but to make our equipment conform to the nature of sound, because the laws of physics are definitely not going to change to suit our convenience, except possibly to do us mischief, as outlined in Murphy's Law.

Okay then, what do we equalize, how do we equalize, and, for that matter, why do we equalize?

Types Of Equalization

Equalization can be divided into two basic types as regards function: **Correctional,** in which the purpose is to correct for faults in some part of the chain which produces deviations from flatness in frequency response, and **Adaptive** in which the response is deliberately caused to deviate from flat in order to improve the operating characteristic of some component in the chain, or to allow optimizing some other parameter, such as noise or distortion.

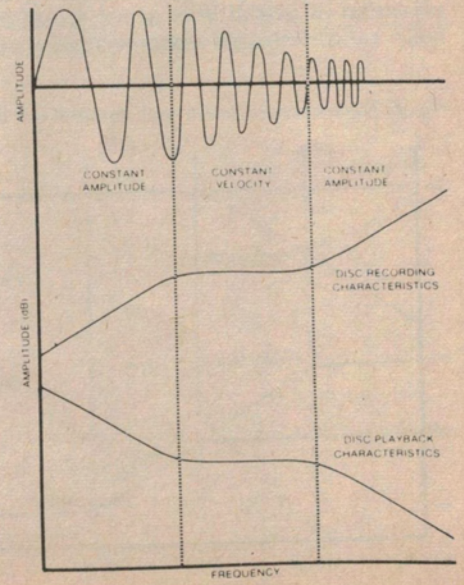
Adaptive Equalization

Several problems occur if we attempt to record or broadcast an audio signal and reproduce it with a flat characteristic from microphone to loudspeaker. As an example let us consider the case of a cutter engraving a disc recording. If a constant voltage is applied to the cutter, this will translate as a constant velcoity of cutting stylus motion. Now, suppose we attempt to record a signal of 1000 Hz and at an amplitude such as to produce a stylus velocity of 10 cm/sec. One cycle will occur in 1 msec and will result in a stylus swing in each direction of 0.025 mm, that is, it will reach maximum displacement in one-quarter cycle which takes 0.25 msec. Now, if we record a signal of 100 Hz at the same velocity, the amount of displacement will be TEN TIMES the 1000 Hz value, or 0.25 mm. 20 Hz would cause a swing of 1.25 mm. Now try to visualize this on a microgroove disc. To allow adequate spacing between grooves, even with no safety factor, would require spacing the grooves at least 2.5 mm apart. If we record over a total of 7 cm of the record surface at 33.3 rpm our maximum possible recording time would be just over 8 minutes. Remember, this assumes no guard space between grooves, which would easily cut this time in half, and assumes we can use the full 7 cm, which would be most unlikely for reasons beyond the scope of an article on equalization. It also assumes we can cut such an amplitude without running into formidable problems with the cutter, and that we can find a pickup which would trace such an amplitude. Remember, too, that 10 cm/sec is not that high a velocity. Clearly, some compromise must be made.

This adaption consists of "equalizing" the recording system so that all frequencies below an agreed upon standard, namely 500 Hz will be recorded at a constant *amplitude* rather than velocity. This results in an attenuation at the rate of 6 dB/octave, and in actual practice this curve is modified at frequencies below about 80 Hz.

Now, it just so happens that magnetic pickups are velocity responsive devices, that is they give equal output voltage for equal stylus velocity. Since our recording was made with constant amplitude below 500 Hz the velocity falls as frequency goes down, and if played back with a magnetic pickup the res-

Fig. 1. Characteristics of phonograph recording and playback processes.



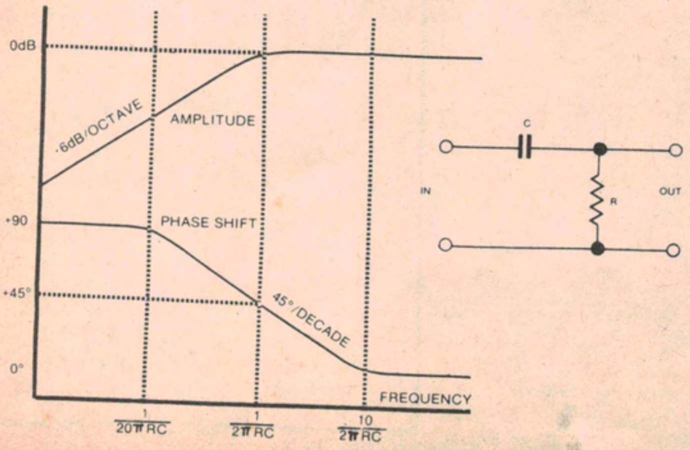
ponse will fall at the rate of 6dB/octave. Clearly, we must now equalize this response by introducing a response characteristic which increases at this rate as frequency is reduced.

The reader who has been following this closely and who has some knowledge of noise (not the kind sometimes called "music", but the other kind) will realize by now that if we continue to record at a constant velocity as frequency rises above our 500 Hz turnover, eventually the point will be reached at which noise generated by surface irregularities in the recording will be equal to or greater than our signal. In addition, noise generated in the pre-amplifier will assume a high level in comparison to the signal. The reader will also realize that this does not have to be, since if we continue to record at a constant *amplitude* we can overcome this noise in much the same way as we did at lower frequencies, i.e.: increase response at the rate of 6 dB/octave as frequency rises. (this is just another way of describing a 6 dB/octave roll-off as frequency drops). It will be appreciated that this could result in stylus velocities beyond the capabilities of the pick-up cartridge, and indeed this is one reason for the use of a modified constant amplitude characteristic: statistical distribution of energy also alleviates some of the potential problems, which makes a fair amount of boost possible.

Another example of this type of equalization is encountered with magnetic tape. A completely loss-free system would show a playback response characteristic which rises at the rate of 6 dB/octave when the tape is recorded with constant flux in the gap, which, in turn, is the result of constant current through the coils of the record head. However because of the tape and head characteristics the response will begin to level off and ultimately drop as frequency rises. The 6 dB/octave slope can be readily corrected (and is) by an appropriate low frequency boost circuit in the playback system, but the high end loss is primarily the result of tape self-demagnetization and playback head losses, with the tape loss characteristic playing a prominent part at lower speeds. This means that inherent tape noise will eventually swamp the signal so that we cannot restore in the playback equalizer, but must do so during recording.

Unlike disc recording the amateur tape recordist is in a position to impose operating conditions during the recording process which conflict with the realities which we have been discussing. Increasing the signal level during the recording process brings the risk of overloading the tape or requires a reduction of overall level which causes deterioration in signal/noise ratio. Boosting response during playback also increases

Fig. 2. Transfer characteristics of simple single pole RC high pass filter.



noise in the active region of the equalizer. As is so often the case in audio work, the end result is a compromise or, with luck, a fine balance between various conflicting requirements.

At the present time it is not my intention to get too involved with equalization circuits; volumes have been written on individual aspects of equalization and doubtless more will be written in the future. However, a brief examination of methods is useful in order to understand the proper use of equipment.

Fig. 2 shows a simple high-pass passive filter, having a first order, or 6 dB/octave slope, along with a normalised frequency and phase response curve. Fig. 3 shows its low-pass counterpart. In figs. 4a and 4b we see these circuits modified to provide low boost and high boost respectively. Notice that the low-boost circuit and its characteristic are derived from the low-pass circuit while the high-boost circuit and its characteristic are derived from the high-pass circuit. It is also clear that a low-pass and a high-cut characteristics are strictly speaking, the same thing, and a high-pass and low-cut are also equivalent. This has led to the mistaken belief which is often encountered even today that bass boost and treble cut are the same thing and vice-versa. An examination of the modified circuit used to provide actual boost immediately shows the fallacy of this notion. *All* equalisers attenuate *all* frequencies equally, except in the relatively narrow region in which boost or cut is required.

Location Of Circuit

The location of an equalizer in a circuit usually involves the reconciliation of several conflicting elements, and is generally determined by the type of circuit (bass boost, treble boost, etc.) nominal signal level, and the operating band.

Since a bass boost circuit involves considerable reduction in mid and high frequency level it is generally desirable to put most of the system gain before the equalizer in order that noise components may be attenuated along with signal. This is also true for high frequency attenuation. However, hum components would then be boosted along with signal; therefore, too much gain will require special attention to design aspects aimed at minimizing hum. Conversely, a high frequency boost circuit should be inserted early enough in the system as to raise signal above the noise of succeeding stages.

Where signal level is fairly high, as with some high output magnetic pickups such as Decca and Empire, our greater concern is amplifier overload, particularly at high frequencies, in stages prior to the equalizer.

If you're starting to get the idea that perhaps equalization, when combined with pre-amplification, can best be accomplished when the equipment is designed for, and associated with, the components with which it is to be used, then you're right on target. Indeed, this is generally considered to be good practice in professional circles. Not only are tape equalizers incorporated into the tape machines with which they are to be used, but turntables may incorporate the required pre-amplifier/equalizer circuits. In over twenty years of audio work I have yet to comprehend the logic behind the common practice of building magnetic inputs into a control unit. But it helps to explain the differences often encountered between results published in equipment reviews and the users' own experience.

Phase

Before moving on to the subject of corrective, or selective equalization some attention should be paid to the matter of *phase*. Hundreds of dollars are often spent on the construction or purchase of, for example, a phono preamp, and great attention paid to noise, channel balance, overload, accuracy of equalization (to within 2 dB) and yet it is seldom realized that,

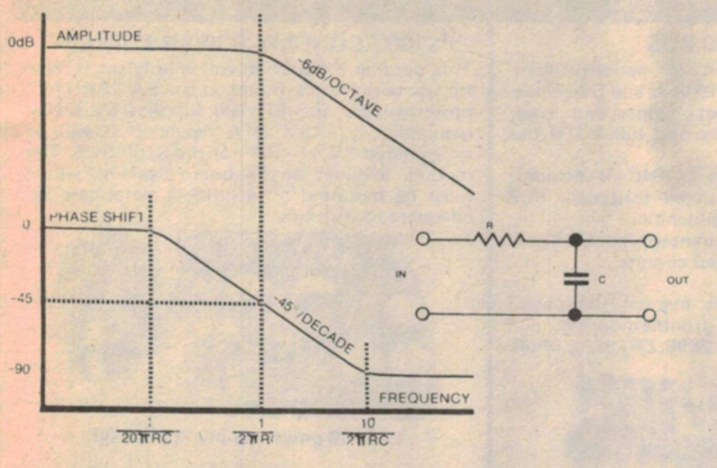


Fig. 3. Transfer characteristics of simple single pole RC low pass filter.

in producing a stereo image, one of the three major factors in directional perception is relative phase. In addition, all matrixed 4-channel systems currently in use utilize a specified phase relationship between channels to encode and decode the additional channels. One common characteristic of all of the impressive demonstrations of quadraphony has been the use of very high quality components. Since equalizers are among the first functions to suffer in making economy cuts in domestic equipment, it's small wonder that quadraphony and even stereo reproduction in the home are often disappointing.

How does this happen? Take another look at the phase and amplitude characteristics in figs 2 and 3. At the turnover frequency, that is the 3 dB down (or up) point, the phase angle has shifted 45° and reaches its ultimate 90° shift a decade away, which also corresponds to the 20 dB point. Now, if common 10% and 20% tolerance components are used in two different equalizers (example, each of a pair of stereo channels), the final curve may indeed we well within 1.5 to 2 dB tolerance in each channel, but if each yields a difference in phase over a broad frequency band of as little as 30° the difference between channels may vary anywhere from 0° to 60°. This is quite considerable in comparison with the 90° shift called for in the parameters of *any* quadraphonic system, even the Dynaco-Haffler passive ambience network. As for stereo perception, although there is much disagreement among authorities as to the ear's sensitivity to phase shift, much of this disagreement involves steady tone conditions and single channel reproduction. When it comes to the perception of a synthesized stereo image as in present day 2-channel systems, as little as 15° has been observed to have a profound effect on imaging, particularly with regard to depth and elevation. In my own experiments I've been able to produce as much as 10 dB channel level difference with no serious effect on the stereo image other than a shift in localization, and yet switching in a simple tone control circuit constructed of standard tolerance parts and with both channels nominally in the flat position will produce a subtle yet real change in image stability and solidity. The indication is quite clear: the precise matching of characteristics between channels is probably of even greater importance than the absolute accuracy of characteristics. This means precision parts and the associated costs.

Corrective Equalization

This might also be described as "discretionary equalization", since it refers to alteration of response in a manner and to a degree completely at the discretion of the operator. In a very real sense the concept of "correct" is irrelevant here. Adjustment is in accordance with the ear's own concept of right and wrong. Accordingly, no hard and fast rules can be laid down nor can definite "how to" instructions be given. However, most of the considerations already outlined apply here, so we may now proceed with an examination of the equipment and techniques available, secure in the knowledge that there is no such thing as magic, and that, rather than use technology to make things better, we can only use it to make things not as bad as they might be.

Uses of Equalizers

Discretionary equalization is used because either we don't think the sound we're getting is correct, or because, correct or not, we don't like it and want to make improvements or produce some special effects. It's something like the photographer who uses a skylight filter to obtain a more realistic colour balance, and the one who uses a red filter to simulate a Martian landscape. Since recordings are made by human beings much of the time, who monitor through loud-speakers with their own characteristics in rooms with their own acoustics, and who apply their own judgement as to what the final sound should be like, it is not surprising that the music lover or audiophile may be in disagreement with the producer from time to time. Perhaps you don't agree that the brasses needed a little extra bite by means of a 7 kHz boost, or you feel that the strings could have been brought more forward and made less disembodied by adding a little mid-range boost. Indeed, perhaps you don't mind a little structural noise of the concert hall, and feel that the producer sacrificed too much bass in order to suppress it. For this kind of correction you will want to use an "equalizer" or tone control of some kind to introduce the appropriate compensation. It may be sufficient to use a simple control circuit such as that outlined in figs. 5 and 6 to get some bass boost, or treble cut. But then there's no way for you to know the exact equalization used in the original recording, and even less chance that a simple circuit could duplicate its mirror image anyway, so at best you can still only adjust it until it sounds better.

If this is not satisfactory, you might try using what is called a "Graphic Equalizer", so called because such a device normally uses slider controls side by side and their settings provide a graphic representation of the response characteristic achieved. With this device you have several controls each of which controls the response over a narrow range of frequencies. It may divide the spectrum into as few as five broad bands or as many as thirty odd very narrow bands. These provide very precise response control indeed, but you still can only adjust response until it sounds right. Just like a simple tone control.

Another type of device is known as a "Parametric Equalizer" because it varies the parameters which define a response characteristic, that is centre frequency, bandwidth, and degree of boost or attenuation. In general, such devices offer fewer choices than a graphic equalizer with regard to the number of centre frequencies which may be operated upon simultaneously. However, in actual use it is generally more flexible largely because of the ability to vary the bandwidth and to choose centre frequencies. In some types it is possible to operate on the same frequency band twice or to operate on two closely spaced frequencies and to combine characteristics to obtain a final response which is completely unobtainable with any other type of component. The parametric equalizer has been widely used in professional work, and is the type of equalizer normally found on each channel of a recording or broadcast production console. There aren't too many in commercial production for consumer use yet, but there is every reason to expect that increasing numbers will be offered to the audiophile. For my money it is the preferred unit of choice for operating on the programme characteristics, provided it is not required to serve other functions.

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The graphic equalizer is probably more familiar to the amateur, since it's been around much longer and several such units have been published as construction articles (See ETI June 1977). The great virtue of this unit especially the 1/3 octave type lies in its ability to notch out or boost one or more very narrow bands of frequencies, making it especially useful in compensating for the irregularities of such components as pickups (RIAA Equalized), loudspeakers, and room

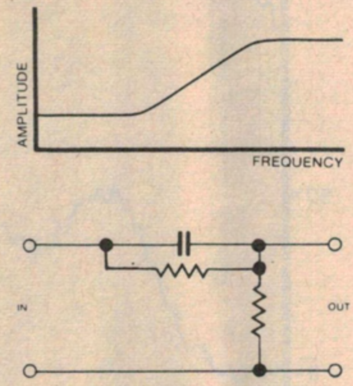


Fig. 4b. High boost circuit and response curve.

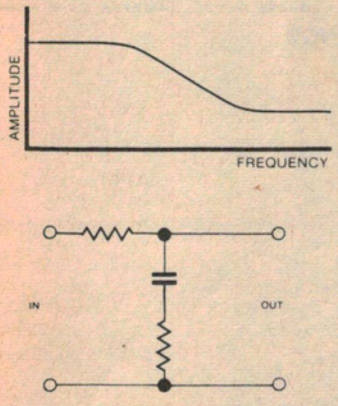


Fig. 4a. Low boost circuit and response curve.

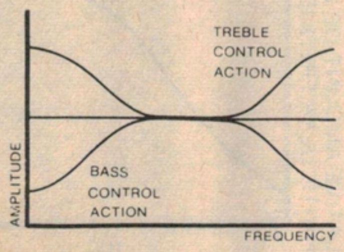


Fig. 6. Range of boost and cut action available from "treble" and "bass" controls.

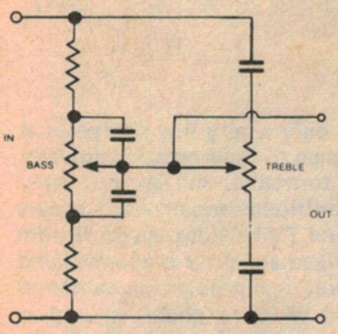


Fig. 5. Typical tone control circuit.

acoustics. Indeed one of its earliest and still common applications among professionals is in equalizing control room/speaker systems. For this purpose, the speakers are each fed with pink noise and the response measured via a calibrated microphone and a real time analyser or other means of measuring response characteristics. The graphic equalizer is inserted in the speaker channel before the power amplifier and adjusted until the desired response (usually flat) is achieved. This procedure is then repeated for each speaker in the system. It should be noted that the response is valid only for the location of the measuring microphone and for the exact acoustical conditions which exist at the measurement.

Two problems arise here, especially for the amateur audiophile. To begin with the previously referred to phase problem can impair the system imaging characteristics. On the other hand it may improve a characteristic which was already deficient because of the phase shifts inherent in the previously uncorrected irregularities. In other words, the assets may exceed the liabilities, both in qualitative and quantitative terms. That's what I mean by an inability to lay down hard and fast rules.

The second problem is more serious. After purchasing or building such an equalizer, especially the graphic type, there is the temptation to use it to correct faults in what is actually a poorly designed system. I can recall one enthusiast who built such a unit from a kit and installed it in a system which was little more than junk. Aside from sounding terrible, it also burned out speakers and destroyed an output transistor.

Why? Well, take a look at fig. 8 which is the representative response and impedance curve which might be expected of a small bookshelf speaker of the sort which promises to outperform speakers three times its size and selling for ten times the price. Although it may boast a response down to 30 Hz, its response at that frequency is down a good 20 dB, and its impedance is equal to the voice coil resistance, around 6 ohms. Now, to flatten the response of such a speaker requires a power increase of 100 times. If, in order to operate at high sound levels, it requires 10 watts of power in its mid-range, 1000 watts would be required. That's quite a lot to demand of a 60 watt amplifier whose ratings are already optimistic, to say nothing of what such power would do to the poor little speaker. Actually, most such equalisers only offer about 12 dB of boost, but if this is combined with a so-called loudness control, it's easy to see the kind of abuse possible.

Another point worth considering is that if you use the control to correct for the equipment faults, you can't use it for programme correction at the same time. If you have 10 dB boost available and you use it all for boosting speaker response, you have nothing left over for programme correction. And the amount of boost available is a function of the equalizer and power amplifier reserve plus speaker handling capacity.

Another consideration is energy distribution. A common assumption is that power levels tend to be about the same at all frequencies. This is not true; as is demonstrated in fig. 9, an energy distribution curve averaged from the results of a variety of studies. It shows that the largest amount of energy in orchestral music is concentrated in the range between about 100 Hz and 500 Hz dropping off rapidly above and more gradually below this band. It should also be remembered that 500 Hz is a common cross-over frequency in 3-way speakers and that with a current trend away from constant resistance networks, many such speakers exhibit high impedances and considerable reactive components in this region, which imposes severe limitations on the power capabilities of many amplifiers. This, in turn, limits the usefulness of many equalizers in this region, especially in the boost mode, where the result is often high distortion and damaged equipment. With a great deal of hard rock, electronic and synthesizer music high frequency energy tends to be considerably greater than with orchestral music. Now, the tweeter of a 60 watt speaker system may be capable of handling typically from 5 to 10 watts of actual power. This is reasonable enough in relationship to the orchestral distribution curve, but an excessive boost in the region handled by the tweeter carries with it the distinct possibility of requiring it to handle anything up to the full output of the amplifier, which may be 60 watts or more. Further, if the tweeter level is padded down (with an L-Pad, I hope) to match the other drivers, this might save them, but much of this power is then dissipated in the pads, and may exceed their ratings.

Then there's the tape recorder. Remember the high frequency boost in the record mode? Well that uses up part of the headroom available. You now have the same problem as with speakers.

Add one final word about high frequency boost. *Every* phonograph pickup has limitations to its trackability. And when it does mistrack it can generate very large high frequency components, which is one reason why it sounds so bad. Moreover groove damage also results, and even if the damaged groove is later tracked with a better pickup there are still quite a lot of extraneous high frequency "garbage" signals generated. If you boost the component signals along with the desired signal, the rest of your equipment doesn't know the difference and will react in the various ways already outlined.

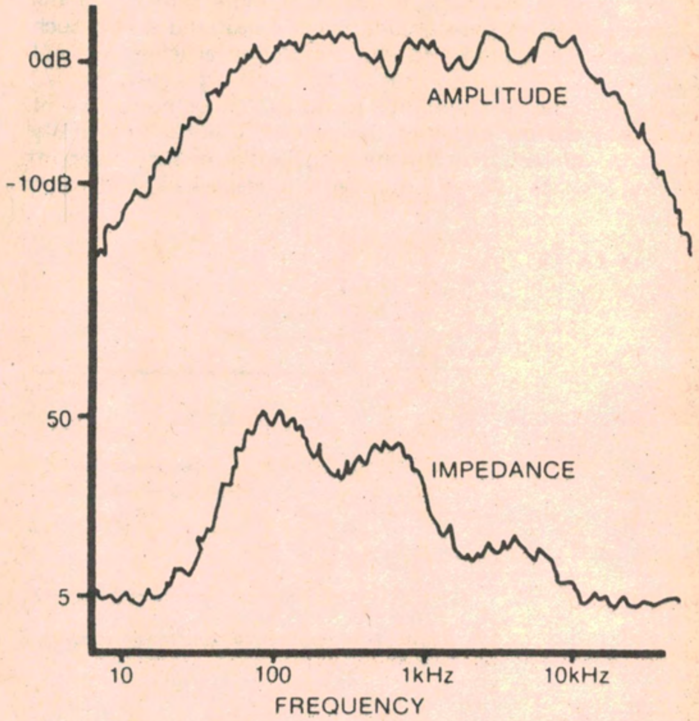
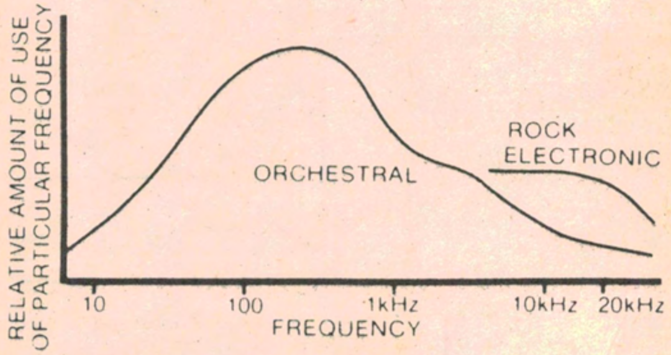


Fig. 8. Typical performance characteristics of low price small size (and high hype) loudspeaker example.

Fig. 9. Energy distribution graph for various types of music.



Circuit Location

The same considerations apply as were outlined with regard to adaptive equalizers; locate at a high enough point in the system to avoid overload problems without boosting noise, and still be functional. Most commercial control units provide a recording output and a monitor return just before the volume control which effectively bypasses all controls including volume and tone, which are used only for monitoring. Installing a graphic or parametric equalizer at this point usually is the most satisfactory as it can then be used for recording and for listening. In general, the most useful point is at the same level as is used for switching. However, if the device is used to correct for speaker/room acoustics the more logical location is immediately before the power amplifier. Unfortunately, this cannot be done with most receivers or integrated amps without opening and modifying the unit. And this is a good argument for the use of separates. But that's another story.

Conclusion

At this point the reader might well wonder what useful purpose is served by these equalizers. In many instances they do more harm than good, but this is largely the result of using them as a substitute for good design. If the performance level sought requires the use of a large Klipschhorn, get a Klipschorn, not a $99.95 super compact speaker special and a magic box. It won't do the job. DO pay attention to the acoustics of the room and the proper placement of a well designed speaker driven by a suitable amplifier. And DO use pickups and other equipment of appropriate quality. Then select the appropriate equalizer if you have use for one, and use it to deal with lesser acoustic problems or to make small alterations to programme quality.

In light of this the reader may be interested to know that my own system uses no equalizers of the discretionary type. A set of large transmission line speakers in a properly treated room provides high level performance through the full audible range with imaging matched by only a very few professional systems. Phono preamps are matched to their own Stanton and Shure pickups each on its own turntable, and are not interchanged. The only need for additional equalization occurs occasionally when taping radio and TV broadcasts and 78 rpm discs. Under those conditions a fixed equalizer is designed and inserted in the line. So far the only real problem encountered is with a particular recording in which a phaser is used in derived quadraphonics - it sweeps the signal around the room and drives the cats crazy.