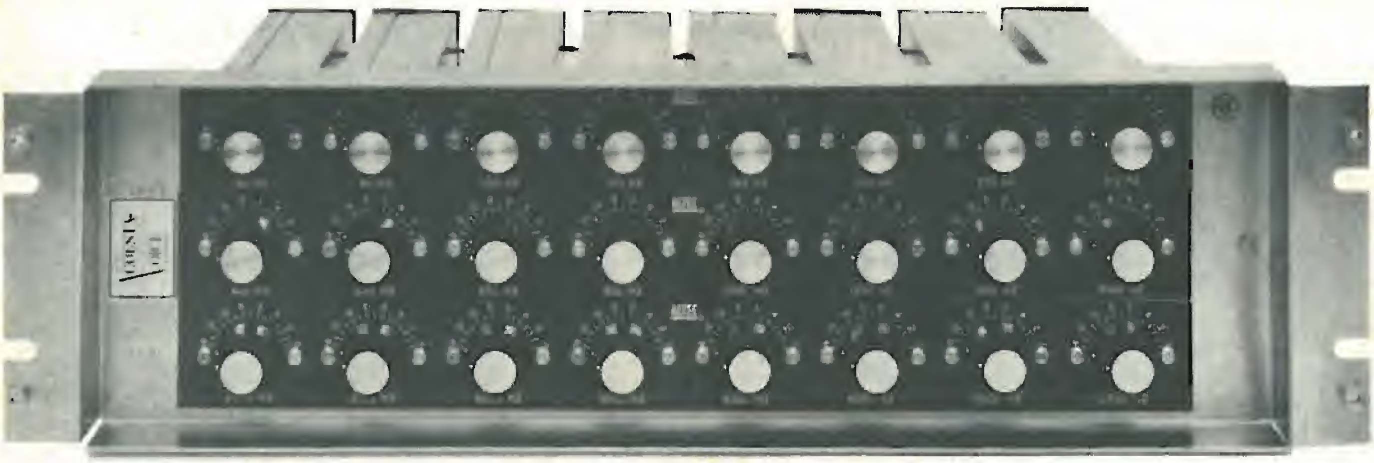


The Altec Lansing 24-filter Acousta-Voice equalizer. Each of the filters can be adjusted in 1-dB steps.

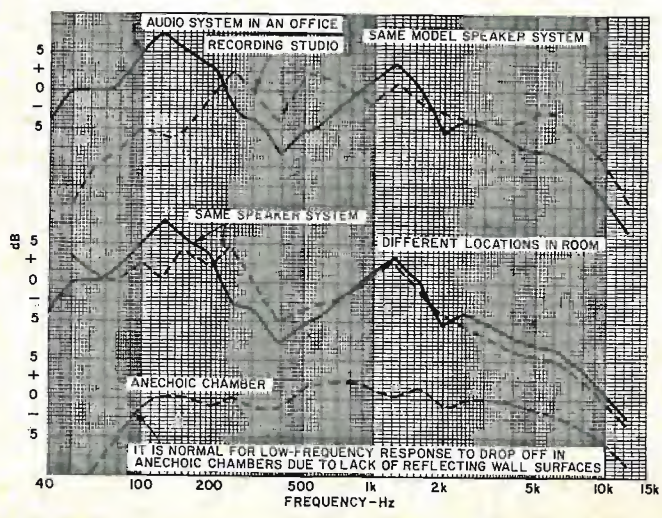
EQUALIZING the Sound System TO MATCH THE ROOM

Here's an equalizer system with 24 band-rejection filters that permits the hi-fi system to be matched to any room—whether an auditorium, concert hall, or at home.

By DON DAVIS and DON PALMQUIST/Altec Lansing (Div. of LTV Ling Altec)

HIGH-FIDELITY amplifiers can be purchased that are capable of maintaining a desired frequency response within ±0.5 dB over the audible range. Recording microphones are available that have a frequency-response accuracy within ±1.0 dB. Phonograph cartridges are available to the home music listener with ±1.0-dB variation in response. In fact, everything proceeds with commendable accuracy and inexorable control until the loudspeaker interfaces the sound system to the room, and the acoustics of the room interfaces the sound to the listener's ears.

Fig. 1. Acoustic response of the same speaker system in various rooms, different locations in the same room (both positions along the same wall but about 12 feet apart), and in an anechoic chamber.



Taking the worst-case variations from recording microphone through the various manufacturing processes and the entire playback chain, the loudspeaker may be fed a signal in the typical high-quality home-music system that is *electrically* accurate within ±2 dB.

In spite of this most impressive technological achievement, the listener actually receives an acoustic signal at his ears that is typically ±10 dB over the range of interest; and in the very best case, ±4 dB. In several cases observed, the signal exceeded ±15 dB in actual rooms with highly regarded high-fidelity equipment.

(Considered subjectively, a change of 3 dB is judged a "just noticeable difference." An increase in level of 10 dB is judged to be about twice as loud. Normal speech levels at a distance of about four feet from the talker measure between 70 and 76 dB.)

The blame for this condition can be almost equally divided between the loudspeaker and the room. This is true mainly because there are no real standards for the performance expected of either. Imagine how much easier the loudspeaker manufacturers' task would be if every listening room had the same shape, size, and absorption, with its loudspeaker placed in a standard location. This would mean that the manufacturer would build such a room at his factory and by placing measuring microphones at the "standard" listening position and the loudspeaker at its "standard" location, he could proceed with the optimum design to provide ±1-dB acoustic amplitude response at the listener's ears.

Until the housing industry decides that the audiophile market is of sufficient size to warrant such special measures and the required time cycle has elapsed to place everyone in his new environment, the problem remains with us.

Traditional Approaches to the Problem

Superficial thinking and knowledge about the acoustical properties of rooms have led some to assert that loudspeakers that exhibit uniform response in an anechoic chamber will continue to do so in a well-behaved acoustic environment outside the anechoic chamber. Such illusions are quickly dashed by a single session with a real-time audio-frequency spectrum analyzer and any high-quality loudspeaker in a typical recording-studio control, cutting, editing or re-mix room. Home living rooms are an even more rewarding environment for such study purposes.

An experimenter will quickly rediscover the over fifty years of observed "room effects" on the performance of a loudspeaker. Bass response will have "holes" in it, thanks to diaphragmatic absorption. This means that some large wall surface in an enclosed space acts as a giant diaphragm and passes the sound at that frequency out of the room by vibrating in resonance with the sound. "Peaks and valleys" attributable to standing waves resulting from the room's dimensions allow a build-up on wavelengths well within the audible range that first peaks the response and then cancels out, depending upon the listener's position in the room. Where the loudspeaker is placed in the room can have a profound influence on the response. For example, in the bass region, response can vary as much as 12 dB from a midroom location to a corner location.

Finally, the room shape and absorption characteristics will have their inexorable effects on establishing the ratio of direct-to-reflected sound at the listener's ears. Fig. 1 illustrates the response of the same loudspeaker system in an anechoic chamber, in different rooms, and at different locations within the same room.

Direct and Reflected Sound

In an anechoic chamber or out-of-doors, as we listen to a loudspeaker at some comfortable normal distance, say 10 or 20 feet, we hear predominantly direct sound. Most of the sound comes to us directly from the loudspeaker. When we go into an enclosed space, such as a living room, we encounter a situation where 10 or 20 feet from a loudspeaker we hear predominantly reflected sound. This means that most of the sound comes to us after first striking a wall or ceiling.

The "deader" a room of a given size, the larger the listening area where direct sound predominates. The "liver" the room is, the smaller is the area where direct sound predominates. Most listeners, whether at home with their hi-fi systems or at the concert hall listening to a live orchestra, sit in what is called the reverberant field where the reflected sound predominates.

Some experimenters have seized upon this aspect of the listening environment and attempted to control the ratio of direct-to-reflected sound in the room. Actually, what is desired is to reproduce in the home environment the same tonal balance the recording hall has in its reverberant field. To do this requires some method of adjusting, in each and every case, the frequency response of the reverberant field at the listener's position in the home to a uniform response, thereby allowing whatever balance the recording has to assert itself.

While different sounds can be produced by reflecting loudspeaker outputs off wall surfaces, accurate reproduction of a concert hall's acoustic environment comes from equalizing the loudspeakers used to produce essentially neutral interaction with the listening room. Then what is heard is a reproduction of the original environment.

Others have attacked the problem from another direction. They take into account the theoretical desirability of an imaginary, omnidirectional, pulsating sphere. However, *all* frequencies must radiate omnidirectionally, and this cannot be accomplished. To further compound the problem, a musical instrument does not radiate omnidirectionally (most have

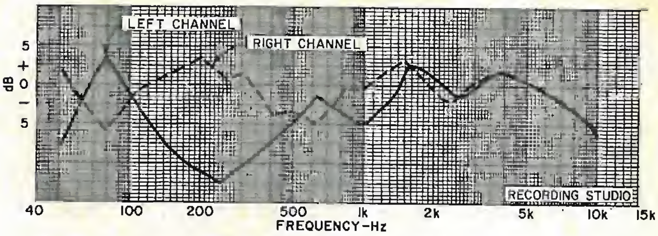


Fig. 2. Response of two identical loudspeakers in a recording-studio editing room prior to equalization. Measurements were taken at the control console operator's position. Differences in response in the 100- to 500-Hz region were dramatic on voices. Voices would appear to move rapidly from one channel to the other.

pronounced polar patterns) and, once in the reverberant field, one has difficulty telling direction in any case.

There Must be an Easier Way

During the past five years an idea was put forth that was first used to correct sound systems that had to operate in difficult acoustic spaces, such as cow-barn arenas, low-cost gymnasiums, or churches with poor acoustics. Both noise control and proper absorption materials had been neglected in many of these places. This idea was to use a highly refined, accurate, and rapid method of equalizing or adjusting the sound system to match the room rather than trying to adjust the room to match the sound system.

The history of sound-system equalization had its early beginnings in the work of Harry Kimball of *MGM*, Ercel Harrison of *Peerless Electrical Products* (a division of *Altec Lansing*), Wayne Rudmose of *Tracor*, and C. P. and C. R. Boner. In 1967 *Altec Lansing* produced the first fully adjustable, fully calibrated, critical-bandwidth, band-rejection equalizer design to permit matching the sound system to the room (see photo). The equalization process developed out of the use of critical-bandwidth filters is called "Acousta-Voicing®."

The equalizer (patent pending) consists of 24 constant-*k*, bridged-*T* band-rejection filters spaced at the standard ⅓ octave center frequencies from 63 Hz to 12,500 Hz. These cross over at their respective "half-pad-loss" points, thereby allowing continuous shaping of a complete spectrum.

The term "Acousta-Voicing" is derived from the practice of voicing and regulating each pipe of an organ after it is installed in the room where it is to be used. The equalizer "tunes" the loudspeakers in the room where they are installed.

The system removes none of the usable program material, but rather brings into equality with the majority of frequencies those special frequencies that the room and sound system together actually tend to over-emphasize. As each over-emphasized tone is brought into equality with all the normal responding tones in the room, the sound quality is vastly improved. Highs and lows are in perfect balance, and the spatial effect in a multi-channel system is startling. (See Fig. 2.) Not only is the sound quality enhanced by being smoothed but now the original spatial relationships that prevailed at the original recording site are reproduced in sharp detail, scaled only by the relationship of the spacing of the recording microphones compared to the spacing of the monitoring loudspeakers.

The long-term effect on recording techniques remains to be seen, but it is possible to conjecture that when home systems are properly equalized, the recording engineer, knowing for the first time what the listener's environment actually is, can safely plan the final recording to sound its best under a "standard" listening situation.

Cost of Voicing

Voicing of a playback or reinforcement sound system to ±1 dB final acoustic response at the listener's ears can be accomplished in 1½ hours per channel by a factory-trained engineer. This time is reduced to 10 to 15 minutes per chan-

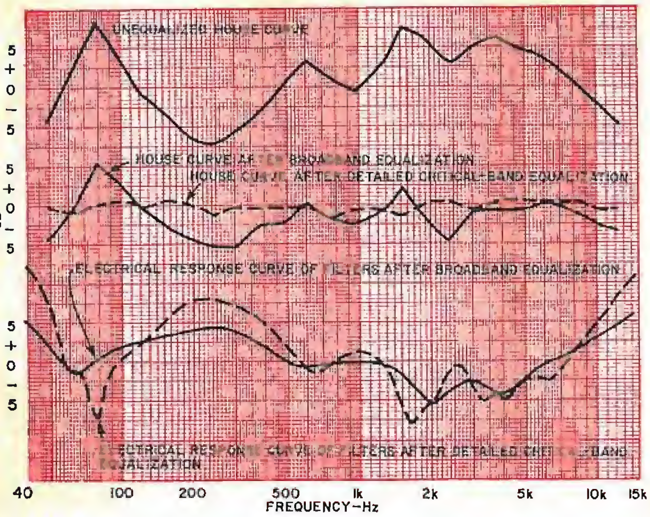


Fig. 3. Attempts to equalize a playback channel with broadband (⅔-octave) filters compared to critical-band (⅓-octave) filters.

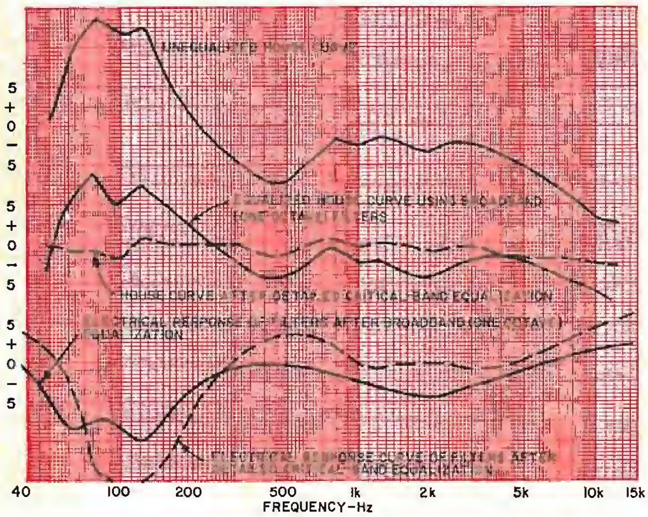


Fig. 4. The inadequacies of broadband equalization are again shown. This is what happened with a corner-mounted loudspeaker facing a wall of undraped floor-to-ceiling windows. The room "seized" the slight bass emphasis and exaggerated it considerably.

nel, after system set-up, with the use of a real-time audio frequency spectrum analyzer. The investment in test equipment is around $10,000 and the cost to the customer is approximately $1,500 per channel and up, depending on the complexity of the system. This is not restrictive for a large commercial installation, but it definitely is for all but the most elaborate home-music systems.

Considerable research has been undertaken to simplify the voicing process in order to reduce the cost for home music systems. Early work attempted to use first five controls with 1½-octave filters, followed by later tests using eight controls with ⅔-octave filters. While these allowed *different* sounds to be produced, they were not sufficiently detailed to actually improve the measured response. Unfortunately, they simply amounted to a more complicated set of tone controls.

It was determined that 24 critical-bandwidth filters spaced at the standard ⅓-octave center frequencies yielded optimum results. Any simpler equalizer could not provide equalization that corrected the problems measured without affecting adjoining frequency regions not requiring correction. See Figs. 3 through 5.

Equalizing in the Home

An intensive effort is being made to reduce the cost of the necessary 24-filter equalizer and to find a very simple but highly accurate and reasonably easy-to-learn tuning method. it is expected that these particular goals will be met shortly.

The high-fidelity dealer will be instrumented for less than $500 to "see" the interaction of the total system—the cartridge, loudspeaker, room—and the change in the house curve with each adjustment of the critical-bandwidth, band-rejection filters. He will be able to guarantee the music system to the customer with ±1 dB acoustic response at the listener's ears. The voicing will be done at the customer's listening room in an hour per channel, and the total cost of the filters will be less than $1,000 for a stereo system.

Fig. 6 is a diagram of where the equalizer is installed in the high-fidelity sound system. In altering a receiver, the technician should be aware that the output from the receiver's preamplifier will see a 600-ohm load and will often be capable of generating only 1 or 2 volts across it with low distortion. Most high-fidelity power amplifiers have input impedances in the region of 100,000 ohms and will require a termination resistor for the equalizer.

There is no need to "build out" the output of the preamp to 600 ohms but care should be taken to be sure to use a low-impedance output. The amplifier should be capable of reaching full output from .1 volt or less in order to meet the required gain overlap of 20 dB (if the preamp only put out 1 volt into 600 ohms).

It is obvious that only first-class equipment will easily adapt to voicing of this type. We suggest 50 watts as the minimum continuous power output to have available. This is because the equalizer is going to allow the system to use enormous power on a frequency-selective basis. This means that a signal can be sending the woofer 30 watts while the mid-range speaker may be receiving power on the order of .3 watt, even though equal sound pressure levels are being generated at both low and middle frequencies at the listener's ears. The 20 dB of equalization makes the difference. ▲

Editor's Note: *The lower-priced equalizer referred to above was demonstrated for the first time several months ago at the Los Angeles Hi-Fi Music Show. The equipment will be delivered shortly to selected hi-fi component dealers whose staffs will be trained to do the tuning.*

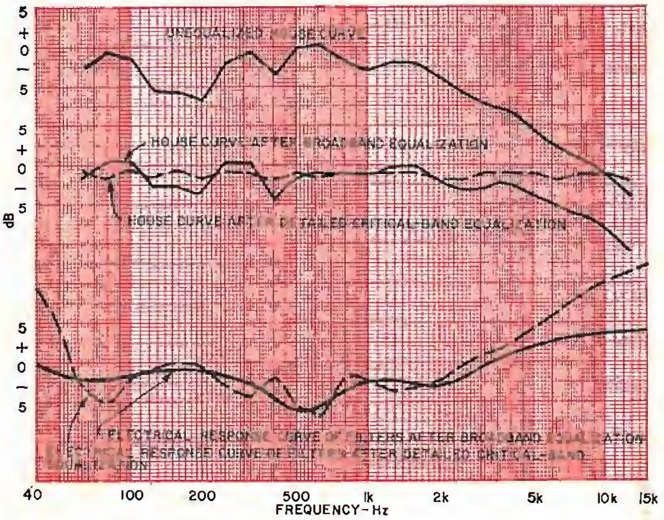


Fig. 5. The response curves of a relatively smooth system in which conventional tone controls can do much to restore overall balance. This type of system is difficult to equalize completely due to close spacing and steep slopes of irregularities.

Fig. 6. Installation of equalizing filters in high-fidelity system.