DOPE FROM HOPE

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SOUND REENFORCEMENT

Back in 1969 Lewis S. Goodfriend, editor of Sound and Vibration, commented on the unintelligible voice announcements at airports, rail stations, etc. And aboard airliners, what could be simply "buckle up" is a tirade which is too loud, usually distorted, always too long, and relatively unintelligible.

Sound Reenforcement, to be adequate, should be louder than the ambient noise, reasonably free of distortion, and be intelligible.

Intelligibility demands a wide frequency response: the t, p, k sounds are carried by frequencies between 4 and 10 kHz.

For churches and lecture halls, an additional criterion would be freedom from listener fatigue. A preacher friend told me he had learned that the attention span of the average member of a congregation is about 15 minutes. (I wonder how the 2-hour harangues of the early Puritans were received; legend has it that sleepers were awakened with a knock on the head).

Thus, if communication is the objective, the sound system should exhibit Quality. Equate quality to freedom from listeners fatigue.

To this end, several criteria are listed.

ELEMENTARY PRINCIPLES

Reenforcement of speech (in churches and auditoriums for example) often suffers from multiple sound sources as well as poor frequency response and distortion. The "lowlevel" system involving a large number of small speakers causes what I refer to as "negative time delay echoes" or "negative echoes" for short. Confusion and listener fatigue are the result.

1. It is good to try to reduce the reenforced sound to one or two loudspeakers (3 if stereophonic) and the loudspeakers should be as far from the listener as is the performer. If the distance is greater by as much as 10 feet or even 20, this is much more tolerable than to have the loudspeakers closer than the person talking.

2. If the distance from performer to microphone is d and from loudspeaker to microphone if D then ideally the performer's voice can be projected farther with reenforcement than unaided by the factor D/d. For example if the performer to microphone distance is 2 feet and the loudspeaker to microphone is 20 feet, then ideally the reenforcement would provide a 10 times increase in distance over the unaided voice. This assumes absolutely "flat" microphone, loudspeaker and room acoustics. Generally speaking the realizable reenforcement will be 3 to 6 dB lower and the distance about half the "ideal" -- in the numerical example the distance gain would be about 5 instead of 10.

3. A suggested "generalized" plan for an auditorium is shown in Figs. 1 & 2. The arrangement offers conformity with principles 1 and 2.

4. It should be obvious that loudspeakers must have adequate power output, low distortion and as nearly a flat response as possible.

5. Irregular shapes and suitable ratio of absorbtion and reflection should be employed in the auditorium.

6. Condenser microphones should be chosen. Omnidirectional condenser microphones offer a more nearly flat response than cardioids, and the flatness offsets the advantage of cardioid polar pattern which is more advertised than real. The cardioid "null" is too narrow to be advantageous when nine tenths of the ambient sound is reverberant. With reasonably flat response from microphones and speakers, and tolerable auditorium acoustics there should be no need for compressors, notch filters and other expensive gadgets.

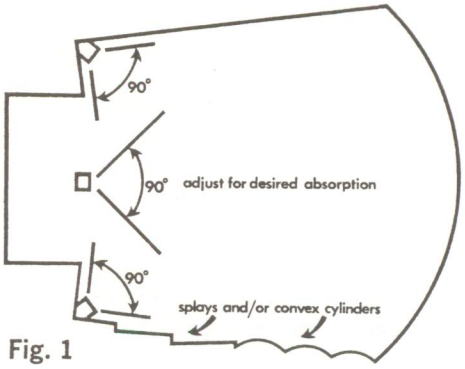
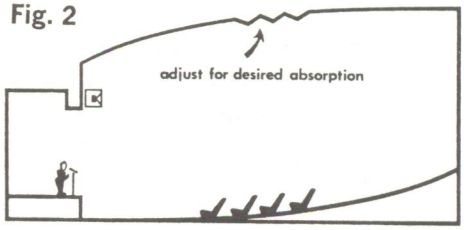


Fig. 1

Fig. 2



MICROPHONES

Microphones with the smoothest response will ultimately be found to offer the highest level of reenforcement before initial effect of feedback (ringing) is noticed. The smoothest microphones are the omnidirectional capacitor (condenser) type.

The cardioid microphone exhibits a polar response much like that of Fig. 3 which depicts an apple sliced open. The "null" (stem area) or low sensitivity solid angle is very small compared to the total solid angle.

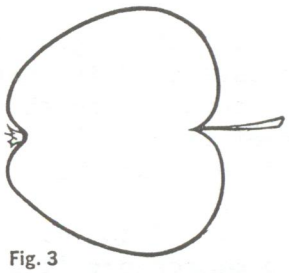


Fig. 3

Experience shows that in an auditorium where reverberation exists at all, the sound from a loudspeaker is bounced off the walls many times and that the sound level is of the order of 10 times what the sound pressure would have been outdoors or in "Free Space". Imagine the reflecting walls as mirrors; one can "see" a myriad of mirror images of the sound source. At which image should one point the stem of the apply? Since 9/10 of the sound is "random" and only 1/10 is directly from the loudspeaker, it certainly wouldn't help, appreciably, in reducing feedback to point the stem of the apple at the loudspeaker. A gain of one dB might be attained, but a gain of 6 dB would probably be possible by using the non-directional (omnidirectional) condenser cicrophone with its more nearly flat response.

This may sound like heresy--"Everybody knows the cardioid is better"--except that they haven't really tried top quality flat-response condensers. We have, and we get 3 to 6 dB higher output before ringing than is specified or possible with other microphones.

A further advantage is the continued smooth response above 8 or 10 even 12 kHz.

Added values are better signal-to-noise ratio, higher output levels permitting less amplifier gain and noise, smaller size, etc.

The feature of smaller size permits better placement. Small microphones, black with black wire, can be hung so as to be above and in front of the performers without being noticed or even seen by the audience. Next to good microphones is good placement, and good condenser microphones are 3/4 inch diameter, permitting them to be placed where they can perform their function without being conspicious.

A recent development of the PZM® microphone appears to offer substantial advantages in public address applications. It is small, wide band and omnidirectional. For speech reenforcement it can be worn on the lapel or as a lavaliere.

EXPERIENCE AND LOUDSPEAKERS

We have participated in some speech reenforcement systems where "you couldn't tell it was on" until the performer got away from the microphone. We were always able to achieve an increase in Sound Pressure Level of at least 14 dB without any incipient tendency to "ring" or exhibit other effects of acoustic feedback.

In planning a sound reenforcement system, consideration should be given to peakfree overall flatness of response, microphone and loudspeaker selection and acoustic environment. These assure balanced sound with minimum feedback. More important for overall intelligibility is minimal distortion.

Distortion is of two kinds, harmonic and modulation. Harmonic distortion is relatively unimportant but modulation distortion can cause loss of intelligilility ranging from muddiness to a severe gravelly quality.

Very few loudspeaker manufacturers offer data on the modulation distortion contained in the output signal. Horn type loudspeakers inherently exhibit only a tenth as much modulation distortion as do direct radiator speakers.

The inherent high efficiency and controlled directivity of horn loudspeakers, along with the low distortion render them the ultimate choice.

ROOM EQUALIZERS
or
NOTCH FILTERS

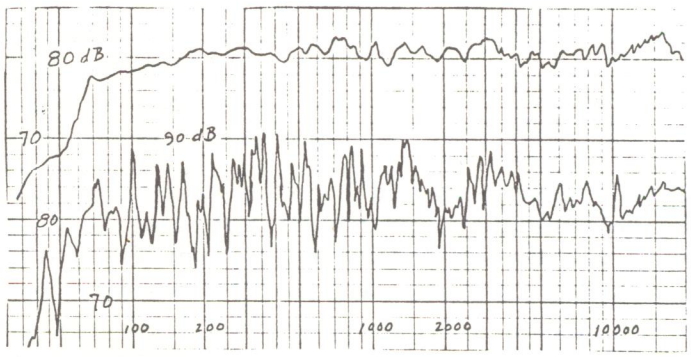
Notch filtering was described by Dr. C.P. Boner in his U.S. Patent 3,256,391. He explored the spectrum of a reenforcing system, increased gain until singing occurs, then "notched down" that particular frequency. The process was repeated over several troublesome peaks.

In our own studio we found some bass resonances and lack of high frequencies. Convex surfaces of 1/8 inch Masonite reduced the bass resonances and increased the "liveness".

Bedell and Kerney of the Bell Telephone laboratories had this to say about room equalization: "If the system including the air path from the loudspeaker to one position in the auditorium is made flat, it will not in general be flat for other positions or other paths in the room". (Symposiumon Auditory Perspective, Trans AIEE January 1934.)

Sound reenforcement in an extremely faulty auditorium may be benefited by electrical equalization, but the same amount of money and effort expended on basic remedies will usually prove advantageous.

To illustrate how a "room" modifies sound, Fig. 4 shows the response of a certain loudspeaker in an anechoic chamber (upper curve) and in a "typical" room (lower curve) for one microphone location. Moving the microphone or listener location changes the response.



Attempting to "smooth" the lower curve would call for perhaps 100 filters of 1/10 octave band width, and as Bedell and Kerney point out, the "equalized" response would apply to only one point in the room -- say, your left ear.

Many years of experience in listening to sound reenforcing systems at home and church teaches me that a few adjustments in the room or hall acoustics can do more than many thousands of dollars spent on equalizers. At home, my wife did not like the room where we put her grand piano. A tapestry hung on the far wall satisfied her.

Modern solid state electronics are remarkably reliable, but the people who operate them are only human. It takes skill to "equalize" a room, and a little common sense applied to the voice room acoustics may well prove more beneficial.

FINALLY

I have tried to keep this treatment of a complicated subject as simple as I can, even while trying to refute some of the popular fallacies. But if a layman is trying to build a new church house or rebuild the audio system in it, and he feels he needs a little hand-holding, telephone us at: 501-777-6751.

Paul W. Klipsch • Klipsch and Associates • Hope, Ark. 71801