The Feedback Loop

—JOHN A. McCULLOCH

The Feedback Loop invites your questions on any subject pertinent to professional audio. Address your queries to THE FEEDBACK LOOP, **db Magazine,** 980 Old Country Road, Plainview, N.Y. 11803. Please enclose a stamped, self-addressed envelope. Mr. McCulloch will answer all letters in this column or by mail.

● A letter from Mr. R. E. Pickett Jr., of WEEW in North Carolina, prompted me to make a review of echo and reverb devices. And . . . the review turned into a search, the search into a project—which is by no means complete. Yet, at this point, it is possible to answer, in part, the questions Mr. Pickett raised.

He asks; "What reverb equipment is primarily in use by stations large and small across the country?". He might also be asking which devices are most like the natural acoustical reverberation. Before we can attempt to answer any questions it is necessary to define our terms, and to determine what effect we want to achieve.

Reverberation is the natural smooth decay of sound in an enclosed space. Larger volumes will have longer periods of decay, and will be more reverberant. Time of decay is not necessarily uniform with frequency. *Echo* is the distinct, time-delayed repetition of the original sound, reduced in amplitude. Amplitude of repetition is not necessarily uniform with frequency.

Lets take echo first, if that is your desired effect. Probably the most simple means by which the repetition of a signal may be obtained is through the introduction of a mechanically delayed signal, reduced in amplitude, into the primary signal. An example would be the use of a tape recorder having separate recording-and-reproduction facilities through which the signal is passed with a portion of the delayed output being reintroduced into the input. Depending upon the speed of the machine and the spacing of the reproducing head from the recording head, the effect will range between rapid multiples and separate and distinct echos.

Several commercial devices using a closed loop of tape and multiple reproduce heads have been manufactured. Because the heads provide the required delays and multiplicity of signals, the problem of system feedback (due to the requirement of introducing the output signal into the input as in the first method described) is eliminated. Good results are possible, but the artificial source of the sound is generally discernable, and is not considered natural. It is, however, a valuable tool which should not be rejected simply as a gimmick.

Reverb is the artificial means of seemingly increasing the size of the recording space. It may be deemed necessary due to a small studio, or more often the use of close miking technique. Ideally, we would wish the effect to be that of the musicians being placed in an acoustical environment proper to the music being performed. To perhaps enable a fuller use of the reverb function (and I am speaking solely of the artificial reverb) we must look at the components and their relationships that make up the natural reverberation of an enclosed volume.

There are three main and inter-related components in acoustical reverberation—*time, amplitude, volume.* For an enclosed space of volume there is an optimum time of reverberation. That time is calculated as the period measured from the cessation of source sound to the point wherein the reflected sounds have been reduced to 1/1,000,000 the source intensity, or −60dB. There may be many time periods found for identical volumes, but different rooms. This is due to the physical properties of the rooms that give different degrees of absorbtion or reflection.

Many papers have been presented giving the *ideal* or optimum period of reverberation for given volumes. W. A. MacNair, in the **Journal of the Acoustical Society of America,*** published figures for reverberation times at 1 kHz *versus* enclosed volume. Subsequent publications contain minor variations of this plot (FIGURE 1) or suggested that a tolerance of this plot, producing a band of acceptable values, be the best representation. For ease of examination we shall use Mr. MacNair's graphic representation.

In the same article, Mr. MacNair also computed the desired rate of decay vs. frequency based upon the nonlinear aspect of our auditory senses (FIGURE 2). This plot was based upon the same data which later led to the Fletcher-Munson curves. It represents the necessary change in decay period from the optimum 1 kHz point to produce a smooth sounding decay. For example, if we have a reverb period of 1 second at 1 kHz, a period of 2 seconds is required at 80 Hz to sound the same. Correcting the room to produce decay **versus** frequency following the curve, will then give a smooth decay, not appreciably varying with frequency, as observed by the listener.

From these two graphs, we may now select an enclosure to our liking, or one which we would like to have had to make a recording. Actual rooms, of course, do differ from the optimum, and if their variations are known, it is not inconceivable that an actual room may be duplicated electro-mechanically. Before we construct the room, let us examine the various means of reverb available in the current market.

To my knowledge there are three methods of producing reverb currently in use. First, and usually most expensive, is the preparation of a special room with highly reflective surfaces, the *reverberation chamber.* (I refuse to call it an *Echo* chamber, as its characteristic is not that of separate and distinct repetitions.) Next are two electro-mechanical methods, each of which has its

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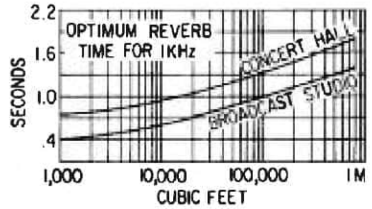

Figure 1. The optimum reverb time for 1 kHz based on the work of Morris and Nixon in the Journal of the ASA, October 1936.

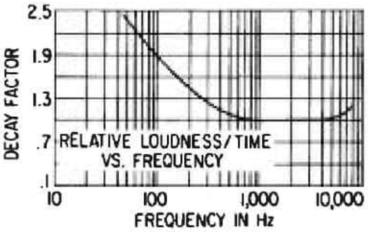

Figure 3. The response curve derived from a small spring.

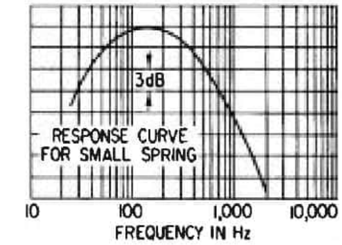

Figure 2. The relative loudness/time versus frequency.

adherants, the *spring* and the *plate*. All three methods, when properly constructed and operated will give excellent simulation. The plate, to be effective is massive, relative to spring devices, and moderately expensive. It is preferred by a great many recording engineers who can not obtain a good chamber. But the widest range in quality, from useless to superb, is covered by the spring device.

Insofar as the spring units are more readily available to us in their basic form, *sans* electronics, and are reasonably inexpensive in that form, we shall look particularly at their natural characteristics. Time decay, and separation between induced signal and first reflection are functions of the spring length. Amplitude decay, frequency response, and level capabilities are functions of the transducers, the material of the spring, and its length. By using a very slow sweep rate it is possible to approximate the characteristic of the unit selected, and thus determine what electronic corrections are necessary to produce a more pleasing, and more nealy optimum reverb.

FIGURE 3 graphs the frequency response of a typical spring unit of the seven-inch variety. Since amplifiers are required to both drive, and retrieve signals from the unit, necessary equalization to provide flat response is easily accomplished. Flat response is **not** what we must seek. A response curve simulating the corrected decay *versus* response curve of FIGURE 2 is the aim. Better units provide adjustable equalization to enable the user to construct his room. Driving the spring unit approximately twice as hard will produce the near simulation of doubling the reverb time. In this manner the correction for decay time may be approximated electronically. While not an absolute method of duplication of existing room conditions, this would allow selective equalization to approximate the varying sizes of rooms.

As far as the use of reverb in broadcasting, the EMI plate, and Fairchild and Fisher springs are the units I most often see. That is not to say these are the only units. In the FCC rules I cannot find a specific ruling governing the use of reverb or echo devices. When I ask engineers who do use these devices, it seems that the rule governing equalizers and limiters is their guide for proof-of-performance. In effect it requires the neutralization of the peculiar action component of the device; that is, no equalization, though the signal passes through. I would assume, and these engineers agree, that if the main signal passes through the device, the reverb function must be defeated and the amplifiers measured. If it only contains the reverb component, it should be eliminated from the circuit.