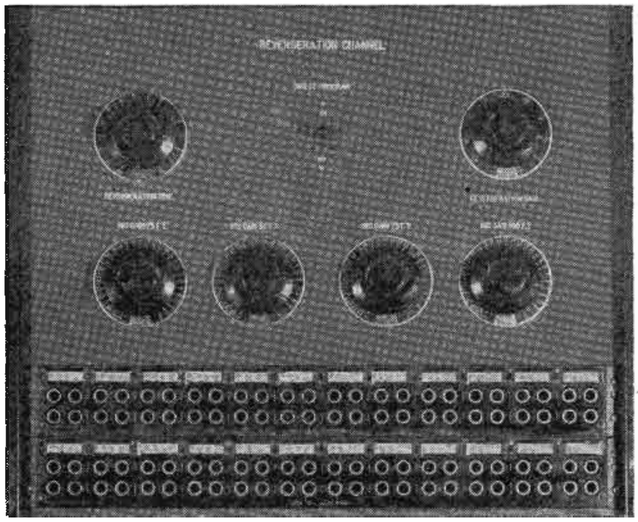
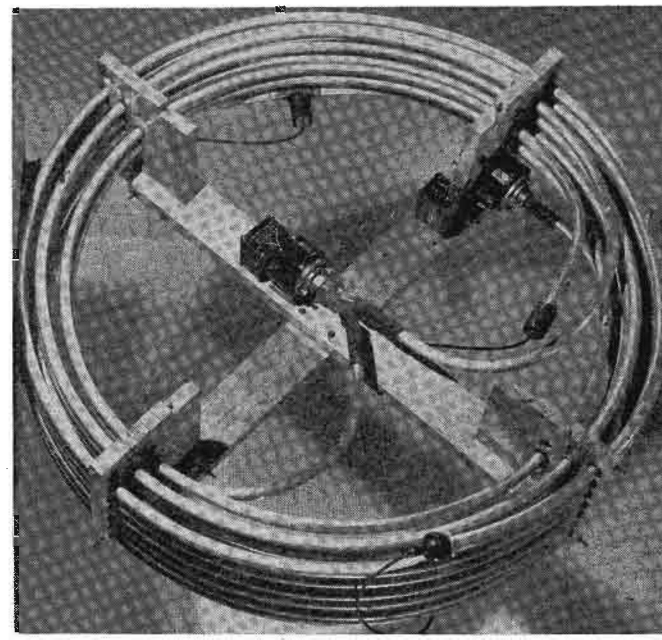
(Right)
Arrangement of the tubing used in the KFI reverberation system.

(Below)
Reverberation panel with reverberation time and gain controls, and microphone gain control for 25', 50', 75' and 100'.





AUDIO PROBLEMS In A-M Broadcasting

An Analysis of Some of the Current Fidelity Problems and Solutions Found at KFI through the Use of a Controlled Reverberation Channel, Automatic Equalizing Amplifier for Transcriptions, and Intermodulation Analyzer.

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ALTHOUGH SIGNIFICANT progress has been achieved in many phases of a-m broadcasting, there still are several factors which have been neglected and unfortunately dismissed by many as unworthy of further effort. This is particularly true of the fidelity problem.

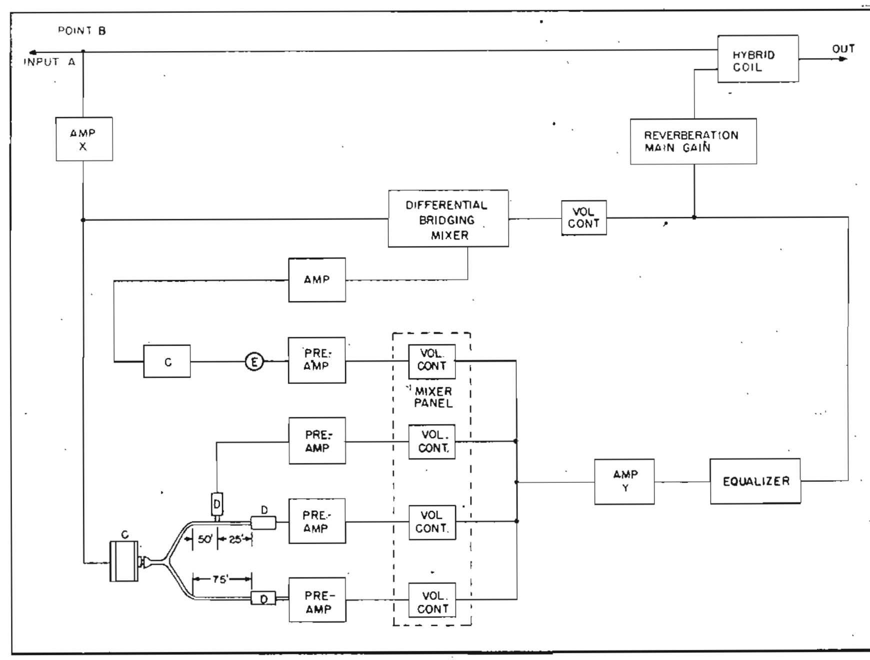
On the abstract idea of fidelity, most engineers have pretty generally accepted as their goal the transferring of music and speech, without a change, from one location to another. This is a comparatively simple process, but still difficult to attain at moderate cost. And today, we are being forced to the conclusion that in a competitive system, even this may not be enough.

The Reverberation Factor

Music and speech, in addition to having fidelity, should be easy to listen to. The human ear seems to be so constructed that, given an adequate frequency response, it is easier to convey intelligence to the brain if there is a small amount of reverberation present either in speech or music. This probably is due to the vibratory character of sound. A reenforcing of the original vibration by a reverberation acting on the ear mechanism, and prolonging the impulses, gives the brain a longer time to absorb the intelligence. Practically everyone who has had the opportunity to listen to music with and without an optimum amount of reverberation has agreed that it was more pleasant to hear; they could actually hear more with less volume on the loudspeaker, and with less effort. In addition, the music and speech had a more pleasing character.

Music is a fine art. Much of its appeal is emotional and depends on small gradations of volume and subtle tonal effects which are produced by harmonics or overtones. The magnitudes of these values are extremely small in some cases, and they are easily lost unless extreme care is taken in pickup and transmission. The fidelity of the electrical part of the system is comparatively easy to attain with currently available equipment, which provides extended frequency response from microphone to loudspeaker. However, no unanimity of opinion has been achieved when fidelity observations were limited to frequency response. The recurring arguments over so-called high fidelity and its acceptance by the public have neglected the reverberation factor, which is possibly a more important ingredient to the pleasure of the listener than an extended-

Figure 1
Reverberation channel used at KFI.



frequency response. Extended frequency response is of course desirable. However other factors such as freedom from distortion and optimum reverberation over the entire frequency range must be present before accurate evaluations of listener response to this question can be made.

It is pretty hard to evaluate the elusive reverberatory effects unless a direct comparison is available. When dealing with listener response to any sound system, it is difficult to reduce the findings to a formula unless all the variables involving that response are under control. It is only recently that any significant advances have been made in acoustic control over an extended-frequency range, and the science has certainly not been reduced to a formula.

Listener response to reverberation has been forceably and sometimes unpleasantly brought to our attention by circumstances. When the transcontinental network was first set up from New York to the coast, there was a noticeable difference in quality between the voices of our announcers and those in New York. The frequency characteristics of our equipment were better than the network, but yet, on a comparison basis, the network announcers' voices sounded rich and full toned, while the voices of our announcers were thin and washed out. Local announcements followed networks with practically no time interval, providing the listener with a direct comparison. This difference was also noted on music which was sometimes even more pronounced. It seems that there was some overall delay effect in the electrical system of the transcontinental network which sounded the same as reverberation and actually enhanced the program. As far as the listener was concerned, this was more important than the frequency response. The magnitudes of the frequency responses involved were 100 to 4,900 cycles for the networks and 50 to 8,000 cycles for the studios.

Reverberation Control

Theoretically, of course, the proper place for control of acoustics is in the theater or studio where the performance is given. In this way the performing artists, either vocal or instrumental, are helped by the reverberation of their own voice or instruments. It is easier for them to control the pitch of their voice or instrument when a considerable amount of reverberation is present. Some artists have gone to the trouble of having concealed microphones installed in the footlights and small loudspeakers placed in the wings of the theater in order to hear themselves sing or play; a simulated single reflection offering appreciable help.

Convex reflecting surfaces have been of immeasurable help in improving studio acoustics, providing an effective control of part of the frequency spectrum. However, standing waves on the bass frequencies which tend to be a function of the size of the room, rather than the shape of the walls, are still a problem.

The application of an electrical device for reverberation control has been found to be quite effective, such a system providing for: (*1*) addition of reverberation; (*2*) control of reverberation in minute degrees; (*3*) control of reverberation in different points in the frequency spectrum.

Most networks and large broadcast stations use an electrical system that provides some of the foregoing features. For instance, the use of a small live room connected to the electrical part of the system through pipes, microphones and loudspeakers is generally used. In general, this is susceptible to resonant peaks which are hard

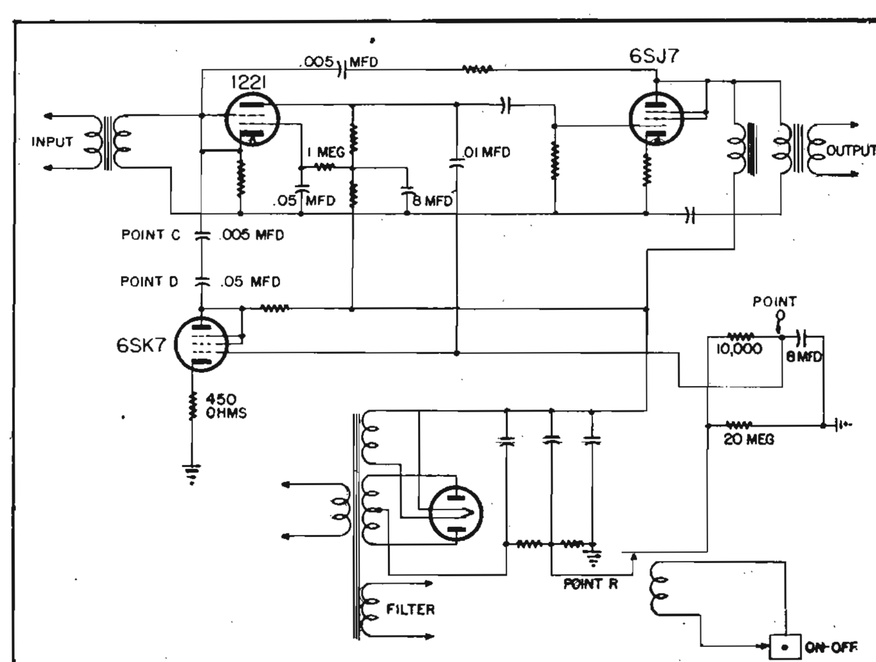


Figure 2
Automatic equalizing playback amplifier, featuring a two-stage preamplifier with equalizing feedback.

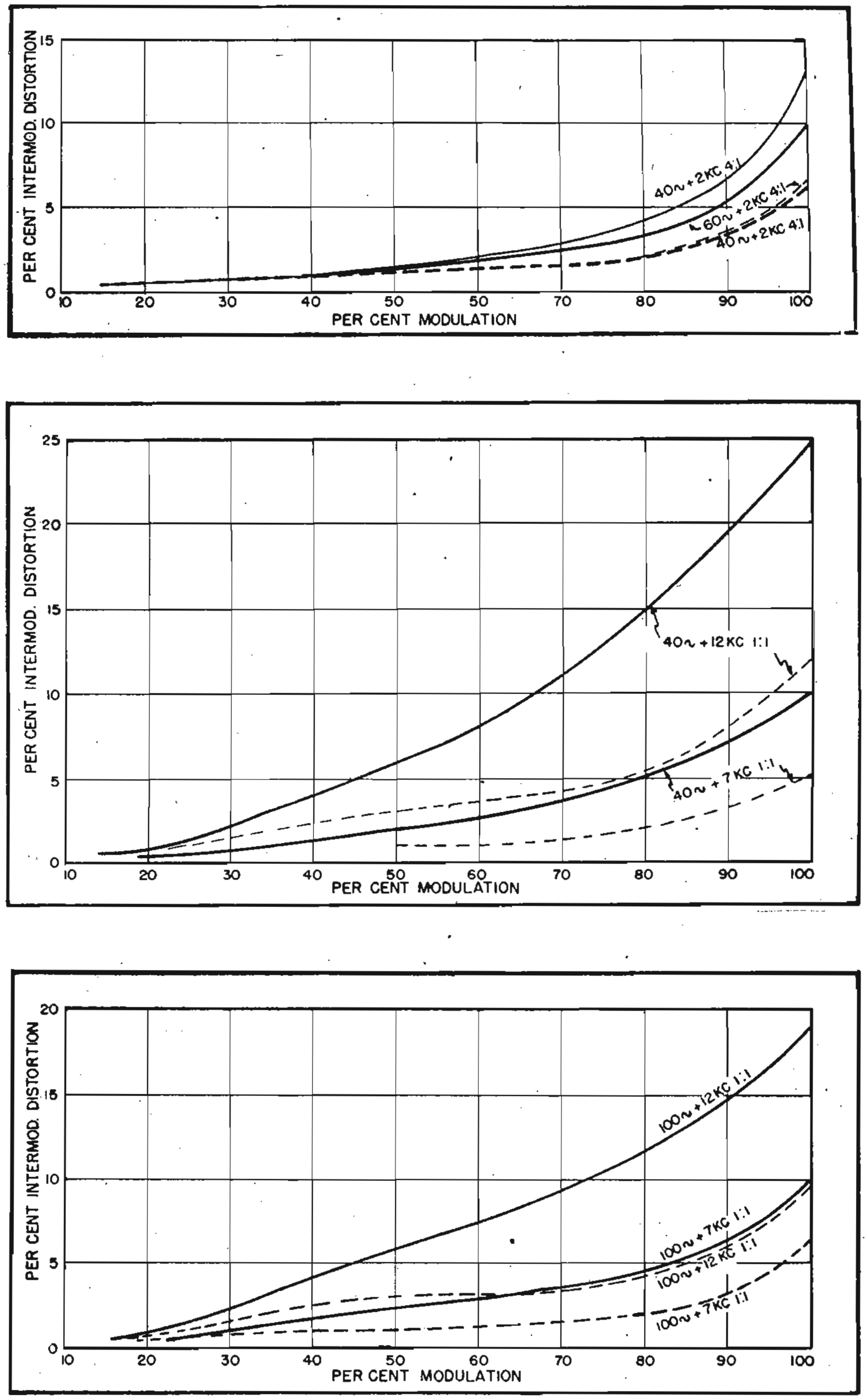


Figure 3

Sample of intermodulation curves of 50-kw transmitter showing reduction in intermodulation distortion effected through use of intermodulation analyzer. The solid line shows the result *before* tuning of final grid circuit and the dashed line shows the *after* result.

to control. A further development of this procedure has been underway at the RCA Laboratories at Princeton, New Jersey, under the direction of Dr. H. F. Olson.

As a result of observations of this system a project[1] to develop a similar device was initiated in 1946. The developed system consisted essentially of four sections of coiled pipe through which sound was transmitted, delayed, fed back and allowed to travel and decay, then mixed with the original sound in order to produce synthetic reverberation.

As this unit was developed and used on actual programs, it was found that the frequency response of the speaker, pipe and microphone unit had to be perfectly flat. A difference of a few db at any one frequency produced a tendency to oscillate and destroy its usefulness. After the frequency response had been made flat, it had to be made controllable both as to the amount of reverberation used, and as to the time period of said reverberation.

Reverberation System Operation

In Figure 1 appears a block diagram of the reverberation channel developed at KFI. Input *A* is bridged at *B* by an amplifier *X*, whose output is connected to *C*, a modified loudspeaker unit.[2] The speaker is connected by a special acoustic device and *Y* connector to 25′, 50′ and 75′ sections of 1″ coiled aluminum pipe. The signal passes through these sections of pipe and is delayed according to the length of pipe, then equalized and fed through hybrid coil to output.

A branch circuit is taken from the output of amplifier *X* and fed through a differential bridge mixer to the speaker, *C*, connected to the 100′ pipe. The output of this is mixed through conventional methods with the outputs of the 25′, 50′, 75′ sections. The volume from each of these acoustic paths can thus be balanced accurately. In order to get true reverberatory effects, it is necessary to return a portion of the output of amplifier *Y* and send it through the 100′ section of pipe, again and again.

This procedure simulates the reverberation effects of any given studio. In addition, the time period of reverberation is controllable with the mixing panel and the amount of reverberation main gain. It was found that extreme care was necessary in equalizing electrical circuits and in coupling speakers and microphones to pipes.

The results, however, seem to justify the somewhat elaborate circuits. We are in the process of learning how to use the device, and in general, we are guided by the effects wanted by our staff musical director and production men. It is remarkable how a small amount of reverberation used in the right spots enhances the music of an orchestra or vocalist.

We intend to incorporate a control so that reverberation can be peaked at certain frequencies. This will allow the operator to compensate for de-

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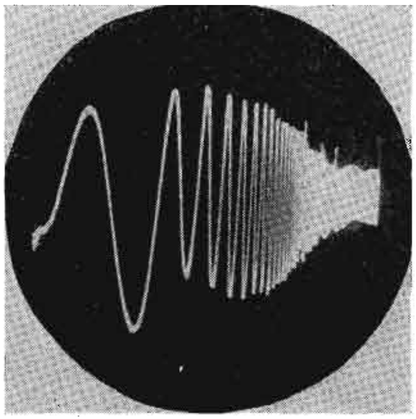
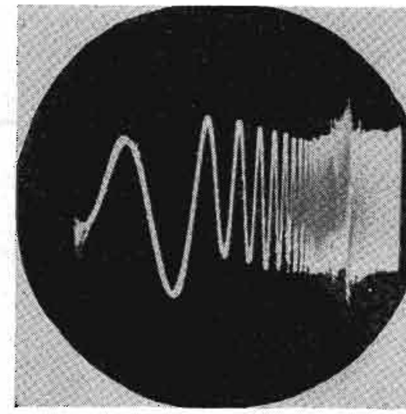
[1]Development by Wayne Johnson, KFI Research and Development Department, which is under the director of George Curran.

[2]W.E. 555.



Figures 4*a* (left) and *b* (right)
At *a* appears an actual photograph of a normal frequency response at inside diameter of 8″ of a 136-line transcription, with *no* equalizing amplifier in circuit. In *b* appears an actual photograph of the frequency response of the same type record with, however, the equalizing amplifier in the circuit.



ficiencies in reverberation which occur in the studio itself.

Delayed Transcription Fidelity Problems

Another fidelity problem has appeared in the delayed transcriptions used to rebroadcast coastal shows, which cannot be presented at originating time at either coast because of the time differentials. For instance, the half-hour Jack Benny show, which originates at 4:00 P.M. on the west coast, is recorded and rebroadcast locally at 9:30 P.M. P.S.T. Thus advertising agency, performers and at-home listeners form a critical and highly articulate audience for our transcription efforts.

The fidelity problem involved the inevitable loss of high frequencies which occurs at the inside start of every transcription. It was particularly noticeable on a half-hour show when switching instantaneously from the outside of one record to the inside of another. Various methods of equalizing the recordings were tried and discarded. Super-imposing diameter equalization on top of the standard NAB preemphasis curve proved impractical because the high frequency peaks overloaded badly and caused a high degree of intermodulation distortion. The actual readings taken on measuring equipment ran as high as 60% intermodulation.

The obvious alternative was to incorporate diameter equalization in the playback instead of the recording. It was also impractical to use any mechanical switch operated by a delicately balanced playback mechanism. Switches are step-by-step affairs and tend to get noisy, especially in low-level circuits, and are thus quite objectionable.

We found that the problem could be solved electrically with a capacitor-resistor combination, by charging it and allowing it to leak off at a rate equal to the rate at which the high frequencies build up as the playback needle moves away from the inside diameter of the recording.

Circuit Operation

The capacitor-resistor combination is placed in a circuit which controls a frequency conscious feedback loop in the pre-amplifier. In actual practice, this changes the playback frequency characteristic of the amplifier from deemphasis to esentially flat at the beginning of the record. This compensates for the loss of the highs at the start of the playback. The needle velocity is low at this diameter, so that needle scratch is essentially the same on both inside and outside diameter of the record.

It will be found by test that most of the high frequencies are lost in the first five minutes of playback and the loss changes logarithmically as the needle travels away from center. Fortunately this is the way a capacitor-resistor combination discharges, and therefore, makes a perfect compensating device.

The circuit, providing an automatic equalizing amplifier, consists of a conventional two-stage preamplifier with equalizing feedback.

When the amplifier is placed in operation and turntable switch turned off, relay *R* is closed, allowing capacitor *C* to charge with voltage from tap on *B* power supply. The voltage at point *O* is applied to the grid of the 6SK7, biasing the tube to cutoff and nullifying the effects of capacitors *B*, *C* and *D* on the frequency characteristic of the first stage. Capacitors *B*, *C* and *D* form a feedback path which is adjusted frequency-wise by capacitor *C* to the deemphasis curve, but only when the 6SK7 tube is not biased to cutoff.

When the relay *R* is operated (by starting switch on the turntable) the charging voltage is lifted from point *O* and capacitor immediately begins discharging through 10,000-ohm and 20-megohm resistors. This slowly changes the bias on 6SK7, thus closing feedback path formed by capacitors *C*, *D* and *B*, and slowly drops the high-frequency characteristic of the amplifier down to the deemphasis playback curve. By the time this has taken place, the playback has moved to a position on the record which calls for

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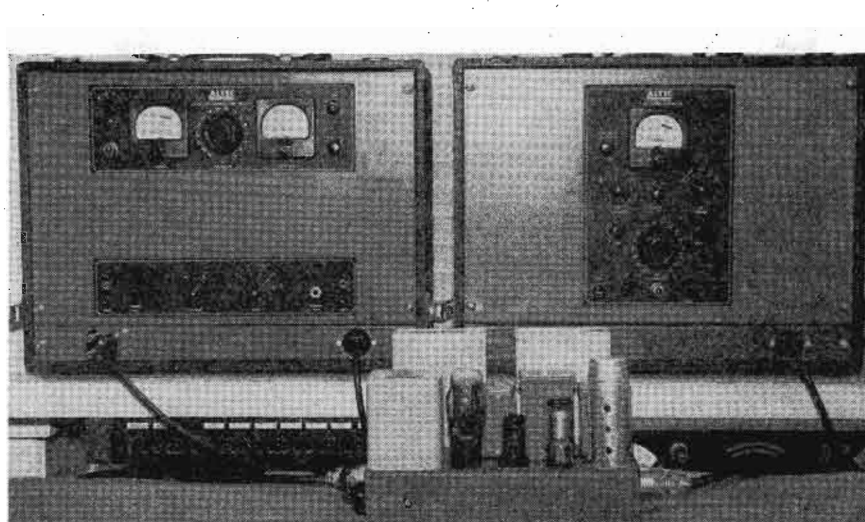
Figure 5
Equalizing amplifier setup with a-f signal generator and intermodulation analyzer test equipment.



Audio Problems

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full effect of the deemphasis curve, a requirement which remains stable until the conclusion of the record.

Since installing these equalizing amplifiers in our playback machines, we have had no complaints which could be traced to changing frequency response of our transcriptions. The inside diameter intermodulation distortion has been reduced to an average of 4%.

Other Transcription Problems

There are other problems in the recording and playback of transcriptions which will shortly be pointed up by their extensive use on f-m. Incidentally we have yet to see a transcription that will comply with the FCC requirements for f-m programs; for instance, noise level of –60 db below program level, frequency response of 30-15,000 cycles and a maximum of 3½% rms harmonic distortion. In addition, we have also noted that intermodulation due to turntable rumble, pickup unbalance, etc., is quite high on the average transcription playback table. Combined noise level of recording table, playback table and record material is generally closer to –45 db below program level rather than –60 db.

Under laboratory conditions we have been able to closely approach the extended frequency range of 30 to 15,000 cycles demanded by f-m, but not with any commercial-type equipment.

Fidelity and the Intermodulation Analyzer

During our development work we have found that the intermodulation analyzer[3] and a-f signal generator[4] are invaluable in making many major fidelity checks. For instance, distortion measurements have been facilitated to such an extent that full potentialities of amplifiers, microphones and loudspeakers could be utilized after a few minutes' test with this equipment.

In our recording department it has been possible to make comparative checks on recording heads by the simple process of recording the two reference frequencies on a given record through each head under test, and playing back and measuring the amount of intermodulation distortion present. Playbacks can be checked by a single recording played and measured by each playback under test. Of course, no one test can determine the goodness of any playback. We have found that some playbacks have excellent frequency characteristics but also a high percentage of intermodulation. By proper centering of armature in the magnetic gap, it has been found possible to reduce this figure to a reasonable level. Compression amplifiers have been checked with this method, and as a result we have been extremely careful in the use of some of them. With some amplifiers, a small adjustment allowed us to use increased com-

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Broadband Radiator

(*Continued from page* 28)

tern of the antenna. The gain over a dipole calculated from this pattern is 3 db above a short dipole.

The center conductor of a 52-ohm coaxial feed cable goes to center of *6* in Figure 3 through a rather long wire which acts as a tuning inductance. This inductance in combination with the capacity across the feed point of the various metal parts and the antenna's self impedance result in 52 ohms feed impedance to match the feed cable. A shorted stub of cable (AN-RG-8/U) is also connected in shunt with the feed point. This stub is effectively $\lambda/4$ long and so appears as a very high impedance at the frequency of operation. It performs no tuning function but acts as a static drain to ground for the upper cone.

In Figure 4 appears a s-w-r curve, which illustrates the match of the antenna to the feed line. As a standing wave ratio less than 2:1 is considered a good match for this type of service it can be seen that this antenna provides an effective match in the 152 to 162-mc range. The antenna is capable of satisfactory operation over a much wider frequency range. Due to shift of the radiation pattern, the gain is slightly less at these extreme frequencies.

Ordinary accumulations of ice and snow have no measurable effect on the performance of the antenna as its broadband characteristics make it insensitive to dimensional changes.

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pression without increase in distortion.

Routine maintenance tests are greatly facilitated by the use of this type equipment. Noise and distortion measurements can be made with a minimum expenditure of time.

Another interesting use of this equipment has been in checking and reducing distortion in the 50-kw transmitter. In a series of checks we were able to effect a reduction of intermodulation distortion, in some cases approximately 13%; Figure 3. This was achieved by a small readjustment of the grid circuit of the final amplifier.

[3]Altec Lansing TI-402.
[4]Altec Lansing TI-401.