BROADCAST STUDIO TECHNIQUES

42 RC

Finished July 23, 1969

NATIONAL RADIO INSTITUTE

ESTABLISHED 1914

WASHINGTON, D. C.

PIONEER HOME STUDY RADIO SCHOOL

STUDY SCHEDULE NO. 42

For each study step, read the assigned pages first at your usual speed, then reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind. Study each other step in this same way.

☐ 1. The Control Room Operator's Job .... Pages 1-2

A brief preview of the equipment used by the control room operator, and the duties of this operator.

☐ 2. Broadcast Studios .... Pages 2-7

The basic principles of program studio design given here are useful to the control room operator of a broadcast station.

☐ 3. Broadcast Microphones .... Pages 7-13

Various types of velocity and dynamic microphones used in program broadcasting are discussed.

☐ 4. Transcription Techniques .... Pages 14-16

The characteristics of commercial transcription pickups and of instantaneous recording are considered.

☐ 5. Remote Broadcasting .... Pages 17-20

Equipment and techniques for remote broadcasts ("nemos") are studied in this section.

☐ 6. Control Room Equipment .... Pages 20-28

The control console and other equipment in the control room used by the control room operator in performing his job are discussed.

☐ 7. Technical Production of Programs .... Pages 28-32

How microphones are placed and used for different types of broadcasts are among the topics considered in this section.

☐ 8. Control Room Operating Practice .... Pages 32-36

Important general considerations in the art and science of control room operating are given.

☐ 9. Answer Lesson Questions.

☐ 10. Start Studying the Next Lesson.

FM1M358 **1958 Edition** Printed in U.S.A.

BROADCAST STUDIO TECHNIQUES

The Control Room Operator's Job

IN this Lesson, we will study the daily routines and duties of the studio control operator (who is often also called the studio engineer), and the studio and control-room equipment with which he will work and must be familiar.

Although we will study the program pick-up and control systems of a.m. broadcast stations, this Lesson applies equally well to f.m. broadcast systems (which we study elsewhere in this Course). The only difference is that f.m. systems are of higher fidelity and capable of greater dynamic range than are a.m. broadcast systems. Because of the large number of a.m. stations with f.m. outlets that use common program pick-up and control equipment, most new stations are designed to handle both, and many older stations have been revised to accommodate both a.m. and f.m.

As we start this Lesson, let us first preview very briefly the control room operator's job and the equipment he uses.

Many programs originate in the regular station studios of the broadcast station. Microphones are set up in the studio in locations where they cover adequately all musical instruments and performers of the production. Sound waves striking the diaphragms of the microphones produce electrical voltages that have the same wave form as the sound waves. These voltages, which are very weak, must be amplified so they can be transmitted by wire lines to the transmitter. The amplifiers used for this purpose are located in the control room. Control of the various microphones is achieved by grouping the switches and volume controls for each on a panel known as a control console. A volume indicator is mounted in an easily seen area on the panel containing the microphone controls.

The control operator places the microphones in the studio where the production director wants them for a particular program. When no production director is present, the control man determines the proper positions for the microphones.

The control operator's primary job is at the control console. He operates the various gain controls of the studio microphones, transcription turntables, and remote lines to blend their respective outputs to produce the desired effect. This procedure will be described in detail later in this Lesson. When a production director is employed by the station, he tells the operator when the desired balance between performers has been achieved; otherwise, this decision is up to the operator.

In any transmission system, there are definite limits to the maximum volume that can be handled and the minimum volume that is adequate for transmission, so the over-all volume of the program is controlled by the control room operator. The volume indicator we mentioned earlier is used to aid the operator in this job. The operator must also operate the switching system to choose the proper studio or incoming line program. A typical control console used for performing these duties will be described in detail in this Lesson.

In general, the requirements for a completely equipped broadcast studio control room audio system may be outlined as:

1. The studio, stage, auditorium, theater, etc., where the performance that is to be broadcast is staged.

2. The required number of microphones for the performance.

3. Amplifiers for stepping-up the electrical outputs of the microphones.

4. Switching and mixing arrangements that let the control man regulate and adjust the output of each microphone used.

5. Incoming and outgoing telephone line terminations on jack panels to permit broadcasts from remote points and to permit flexibility in operation of control room equipment.

There are two general types of control room design and layout. Key stations of a network of broadcast stations, and some of the larger independent stations, have individual control rooms for each studio. A "master control room" is also provided where the output of any studio may be distributed to any one of various output lines and to the transmitter.

The great majority of broadcast stations, however, have only one centrally located control room from which all studios are supervised. Switching and mixing arrangements are located at one central control console and controlled by one operator.

Broadcast Studios

We will begin our study of broadcasting systems with the studios where many of the "live" shows of a station originate. Although the control room operator may never have occasion to design a broadcast studio, an understanding of the basic principles of studio design and construction is a valuable aid to an operator in performing his job. A control room operator, for example, frequently has to "set up" a studio (that is, arrange the microphones in a studio for the best pickup of the program). A knowledge of studio characteristics is helpful in this job.

STUDIO PLACEMENT

To help the operator perform his duties, the studios are generally grouped around the master control room as shown in the typical example of Fig. 1. Notice that the control operator is able to look directly into each studio.

STUDIO ACOUSTICS

Broadcast studios are, or should be, designed to permit an accurate and pleasing reproduction of the program being presented. To do so, they must meet the following general requirements:

1. Freedom from external noise.
2. Diffusion of sound, with uniform distribution of sound energy throughout the microphone area.
3. Freedom from resonance effects.
4. Reduction of reverberation so that

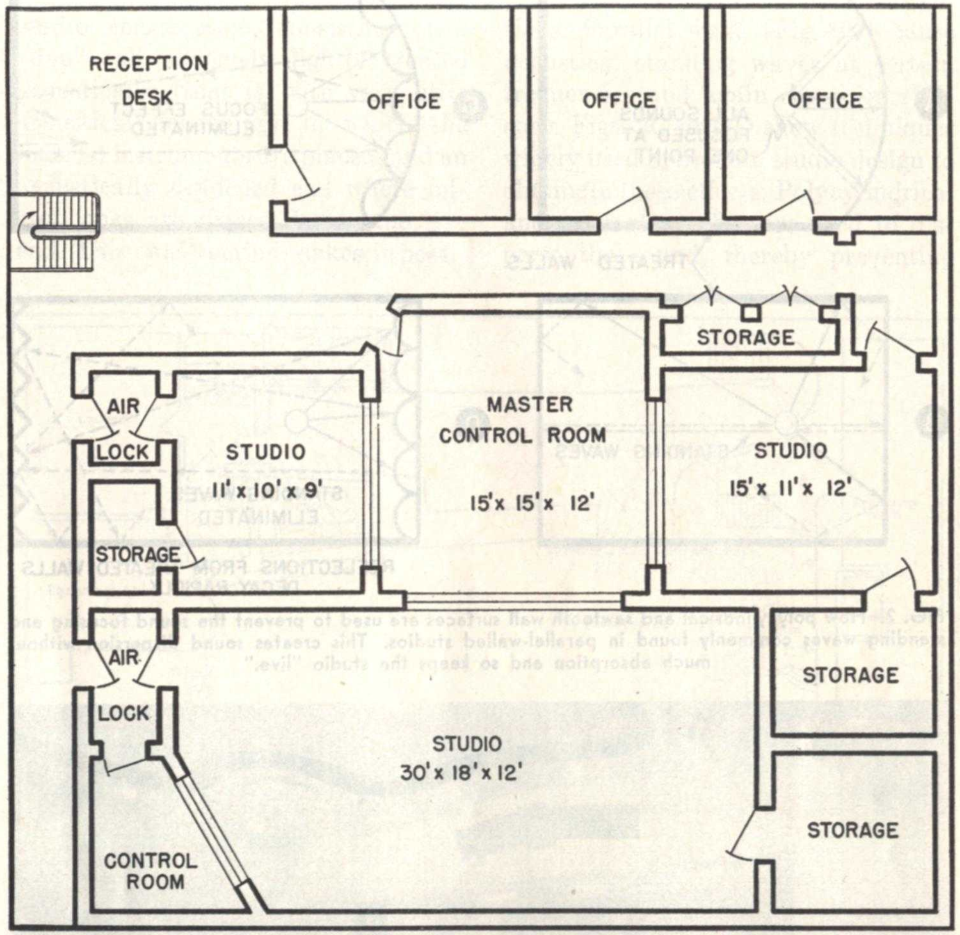


FIG. 1. Typical Studio Layout for a small broadcast station. This shows how the master control room is arranged so that observation in all studios is possible. Note also that air locks with double entrance doors frequently are used for studios. This minimizes the possibility that extraneous noises will affect the program being presented.

excessive overlapping of sound energy does not occur.

5. Sufficient reverberation to make musical overtones audible.

Sound-Absorbing Studios. In the earliest days of broadcasting, when "studios" were simply ordinary rooms not isolated or acoustically treated in any way, the most troublesome problems were quite naturally noise and echoes. The first steps taken in studio design were to treat the walls with a sound-absorbing material to prevent echoes, and to create isolation from external noise by eliminating windows from the room so that the acoustical treatment covered all the wall surface. This type of studio construction, though it eliminated echoes and noises, was acoustically "dead." The slight amount of reverberation necessary for emphasis of musical overtones was also eliminated. Many older studios of this sort, however, are still in use.

Live-end, Dead-end. This difficulty led to the "live-end, dead-end" type of

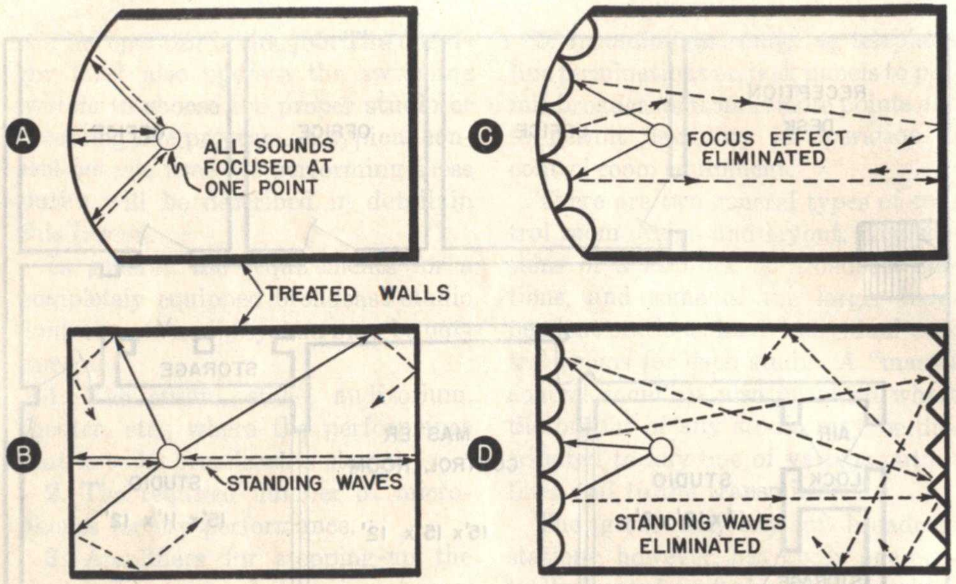


FIG. 2. How polycylindrical and sawtooth wall surfaces are used to prevent the sound focusing and standing waves commonly found in parallel-walled studios. This creates sound dispersion without much absorption and so keeps the studio "live."

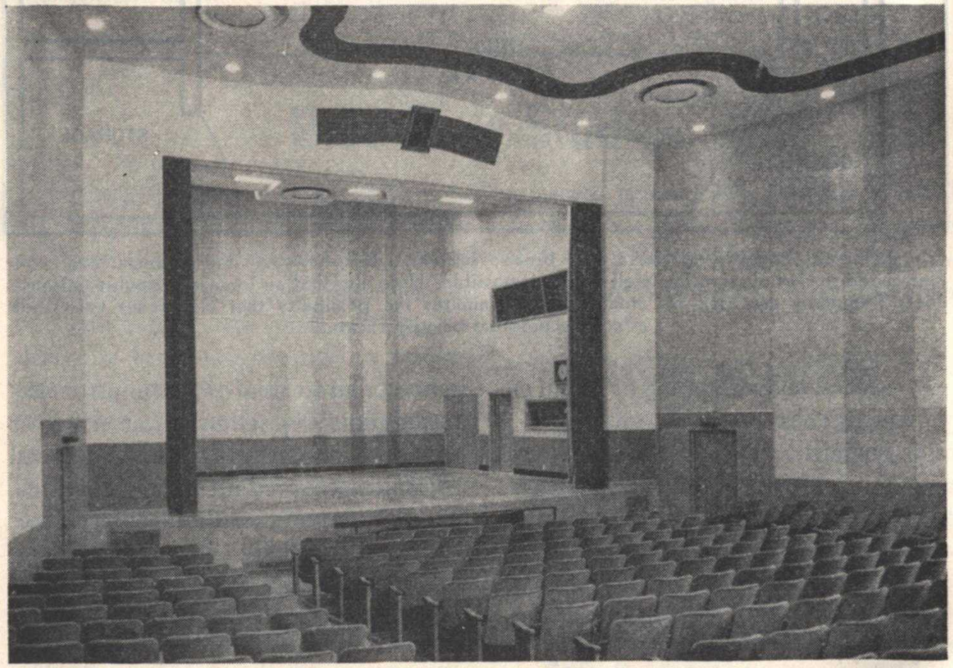


FIG. 3. A typical large "live-end, dead-end" studio. The program is presented on the stage where there is very little sound absorption, and sawtooth wall panels are used for sound dispersion. The audience is seated in the "dead" portion of the studio where sound absorbing material is used on the walls. (The audience itself is sound absorbing.) The seats are covered on the front for further sound absorption.

studio construction, consisting of a "live" end relatively lightly treated acoustically (that is, with very little acoustical deadening) in which the musical instruments are placed, and an acoustically deadened end where microphones are placed facing the live end. This construction makes it possible to obtain diffuse sound distribution and brilliancy of musical overtones without bothersome reflections from side and rear walls where the microphone is placed. The live end is constructed with no parallel walls, thus eliminating reflections and yet achieving the liveness necessary for good musical pick-ups.

Figs. 2A and 2B show two of the basic problems encountered in studio design. In a studio shaped like the one in A, the sound is concentrated at one point and may cause undesirable reflections. Parallel walls (Fig. 2B) cause acoustical standing waves at certain frequencies and again cause interference. Figs. 2C and D show techniques widely used in modern studio design to eliminate these effects. Polycylindrical and sawtooth baffles are used to disperse the sound, thereby preventing sound focusing and standing waves. This can be done without absorbing the sound, so the studio is still "alive."

An example of a "live-end, dead-end" studio is shown in Fig. 3. It uses sawtooth walls for sound dispersion in the "live end" (the stage) and has sound-absorbent walls in the "dead end" (where the audience is seated).

The most modern studios of network stations have provisions to change the reverberation time of the studios to suit different types of programs. Fig. 4 shows the layout of a CBS studio in

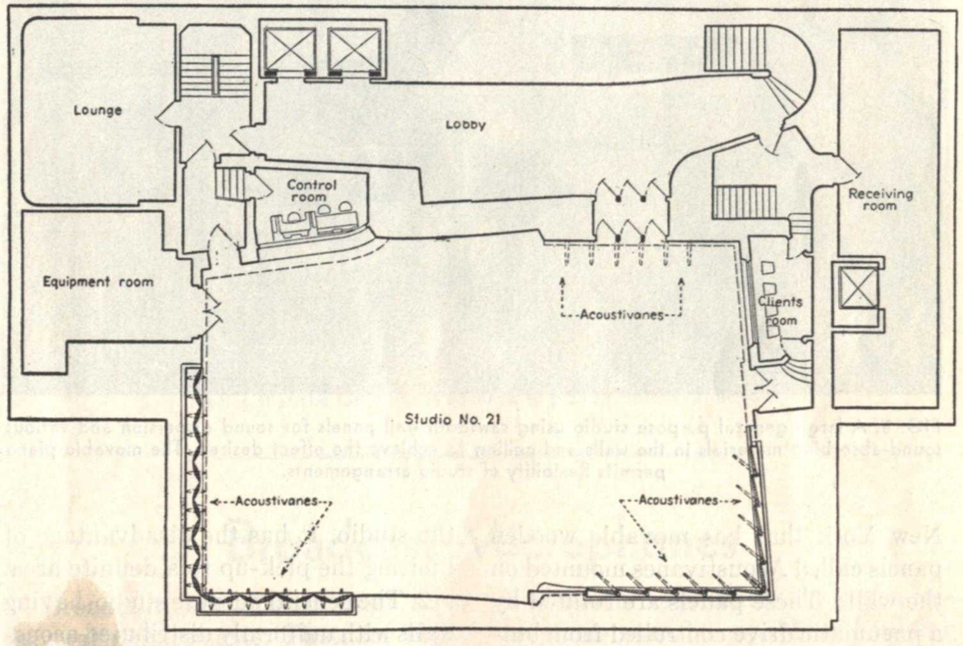


FIG. 4. Layout of a large studio with its control room, equipment room, lounge, etc. The Acoustivanes are large rotatable wood panels that are used to control the acoustic characteristics of the studio.

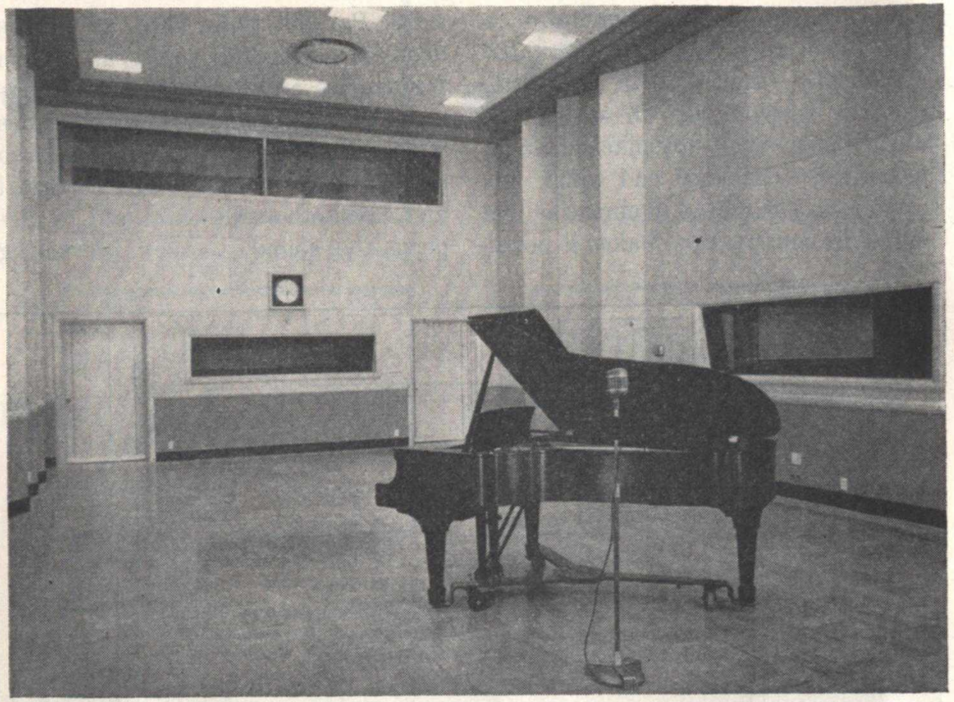


FIG. 5. A large general purpose studio using sawtooth wall panels for sound dispersion and various sound-absorbing materials in the walls and ceiling to achieve the effect desired. The movable piano permits flexibility of studio arrangements.

New York that has movable wooden panels called Acoustivanes mounted on the walls. These panels are rotated by a pneumatic drive controlled from buttons in the control room, and may be moved to any angle desired. The wall behind the panels is of sound absorbent material; when it is exposed, the reverberation time is greatly reduced.

Smaller stations do not have such elaborate studios. In general, modern studios of the typical broadcast station are of two types:

1. The live-end, dead-end type just mentioned, in which the musicians are placed in the live end, and microphones are placed in the "microphone area" in the dead end. Such a studio has the advantage of retaining a definite reverberation time regardless of the size of the studio audience in the dead end of the studio. It has the disadvantage of limiting the pick-up to a definite area.

2. The general purpose studio, having walls with uniformly distributed acoustic treatment or panels of different types of acoustical elements to achieve a desired condition. Many materials of varying acoustical properties are available, and almost any desired acoustic characteristic can be secured by using the right amounts or adjusting the orientation of these materials in a studio. Such a studio has the advantage of unlimited pick-up area, but has the disadvantage of being affected by the size of the studio audience, since the reverberation time when the studio is vacant is much different from what it is when the room is occupied by a large group of people. Figs. 5 and 6 show two typical general purpose studios.

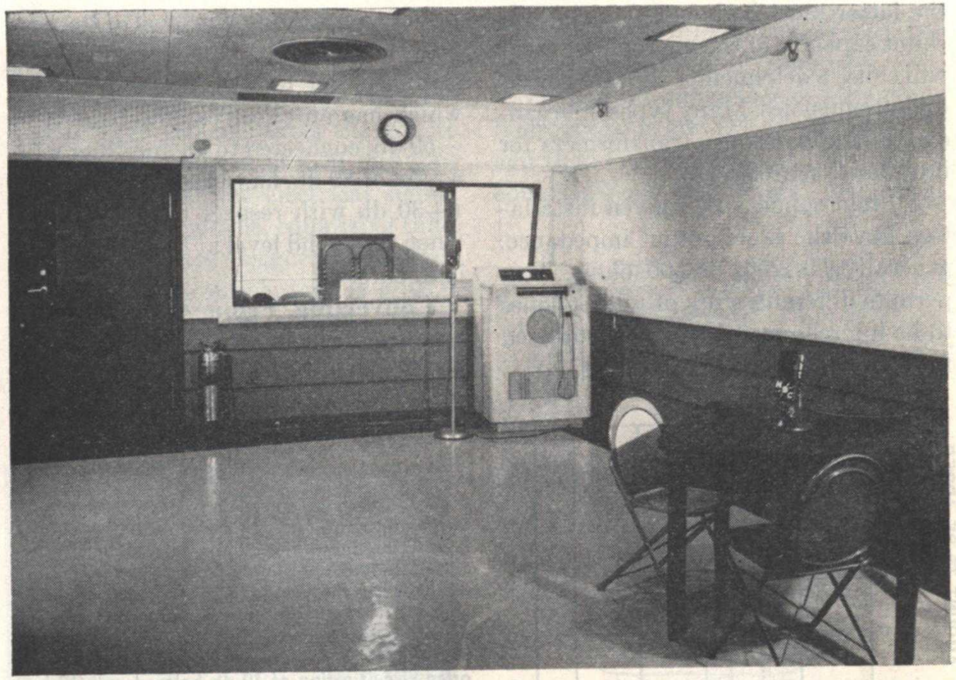


FIG. 6. A small general purpose studio. The audience is placed outside the studio in this case. Notice that the table top is covered to prevent sound reflection.

Broadcast Microphones

Broadcast microphones must meet several requirements. First, for high-fidelity operation, the microphone must have a reasonably flat frequency response from 30 to 15,000 cycles. For a.m. stations, where adjacent-channel interference, receiver limitations, and transmission line characteristics limit the usable frequency response, a reproduction of frequencies between 30 and 8,000 cycles is considered high quality. However, since frequency-modulation broadcasting (to be studied later) requires high-fidelity reproduction, a.m. installations that also have f.m. outlets generally require full fidelity in microphones and in all other audio equipment.

A second requirement is that a microphone must be capable of handling a large range in sound pressures without distortion and with a minimum of noise. The dynamic range of a symphony orchestra, for example, is about 70 db. This means the sound pressure on the microphone diaphragm varies over a 3000 to 1 ratio. At the high sound pressures, there must be a minimum of distortion; at low sound pressures, the inherent noise in the microphone must be much lower than the signal (—40 db in a.m. broadcasting) for satisfactory operation.

Since microphone preamplifiers are generally at some distance from the microphones, the output impedance of the latter must be low. Otherwise, the shunt capacity of the microphone cable will cause a detrimental shunting of the high frequencies. Many types of broadcast microphones use transformers for impedance matching.

All microphones at a given installation have the same output impedance, generally 50, 250, or 500 ohms. This permits different types of microphones to be interchanged freely without need for adjusting impedance values.

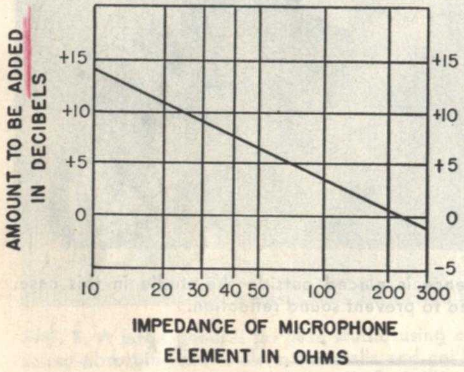


FIG. 7. Chart for converting "open circuit voltage" rating of a microphone to "effective output level." This shows the number of decibels to be added to the open circuit rating. Note that it depends on the impedance of the microphone element (not the output impedance from the matching transformer). The conversion for a 50-ohm dynamic microphone is + 7db, thus an open circuit rating of 70 db below 1 volt/10 bar is equal to an effective level of —63 dbm.

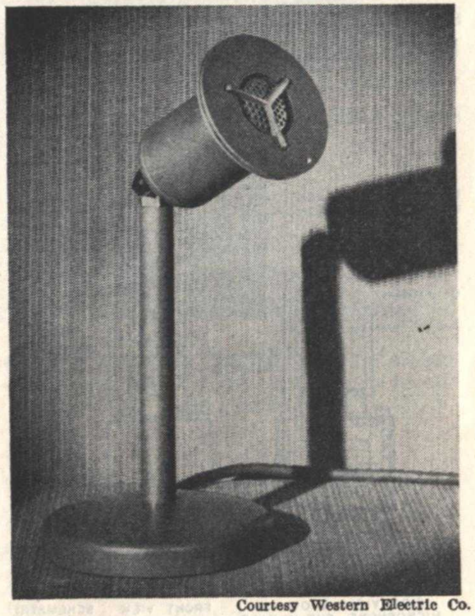
Output Level. The output level of a microphone can be rated by different methods. One method is to indicate the output voltage in db with respect to 1 volt when the sound pressure is 10 acoustical bars and the microphone output is connected to an open circuit. The Western Electric cardioid microphone, for example, has an output of —64 db with respect to 1 volt under these conditions.

Normally, however, a microphone is terminated in an impedance equal to its own, in which case the output voltage is half the open-circuit voltage. The power in the load can then be referred to the standard one milliwatt to obtain a rating known as the "effective output level." This rating is very useful in that the effective output level of a microphone can be added to that of an amplifier to determine the level of the output signal. For example, a system consisting of an RCA 77-D microphone, which has an effective output level of —59 db, connected to an amplifier with a gain of 89 db, has an output of 1 watt (+30 db with respect to 1 milliwatt) when the sound level at the microphone is 10 bars.

Converting From One Level to Another. The effective output level of a microphone depends not only on the open-circuit voltage rating of the microphone but also on the impedance of the microphone element. Fig. 7 is a chart that shows how many db must be added to the open-circuit voltage rating to obtain the effective level. For example, the 633A dynamic microphone in Fig. 8 has an open-circuit voltage rating of —70 db and an impedance of 20 ohms. Tracing up from the 20-ohm mark on the horizontal scale, we find that 11 db must be added to the open-circuit rating to get the effective output level. This latter is —70 + 11 = —59 dbm (decibels below 1 milliwatt).

We will generally use the effective value in giving the output ratings of various types of microphones.

Dynamic Microphone. A cut-away view of the "eight ball" type of dynamic microphone widely used in broadcasting is shown in Fig. 9. The sound wave entering the top of the "mike" through an acoustic screen and a protective screen causes the coil to vibrate in the magnetic field and thus produces the output electrical signal. The air passage, baffle, equalizing tube, and air outlet are designed so that the pressure on the diaphragm is greater at low frequencies than at the middle range of frequencies. This compensation is needed because, without it, the coil output voltage (being proportional

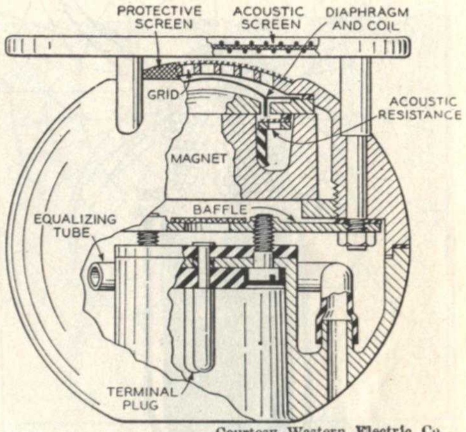


Courtesy Western Electric Co.

FIG. 8. With the disc-shaped baffle applied, the 633A "salt shaker" dynamic microphone becomes a directional unit, and by means of the convenient universal joint it may be literally aimed at the sound source to be picked up, effectively shutting out sounds from other directions. Without the disc-baffle, this unusual little microphone becomes a non-directional unit by returning it to a vertical position.

to the rate of mechanical change of the coil) would be proportional not only to the amplitude but also to the frequency of the sound wave, and would therefore decrease at low frequencies. The compensation keeps the frequency response of a dynamic microphone reasonably uniform from 50 to 10,000 cycles.

The output impedance of a dynamic microphone is low (about 50 ohms). It can therefore be connected directly to comparatively long "mike" lines with-



Courtesy Western Electric Co

FIG. 9. Cross-sectional view of the Western Electric "Eight Ball" dynamic microphone.

out undue attenuation of the high frequencies.

The output level of a dynamic microphone is also quite low, about —55 db (effective). This means that carefully shielded microphone cables must be used to avoid picking up stray electrical fields that would cause undesired noises.

The dynamic microphone is very rugged and can be handled gently while it is "live" without producing stray noise or damage to the unit. It is, therefore, widely used for "remotes."

The dynamic microphone in Fig. 9 is non-directional. As you can see from its construction, sounds from any horizontal direction will affect the diaphragm equally well.

The Ribbon Microphone. This microphone, sometimes called the "velocity" or "pressure gradient" type, is perhaps the most popular in use in broadcast stations today. Its operating principle is that the incoming sound wave causes a thin corrugated ribbon of metal to move back and forth in a magnetic field, producing a voltage between the ends of the ribbon.

Courtesy RCA

FIG. 10. The RCA 77 polydirectional microphone. It is a ribbon microphone with one portion of the ribbon "velocity-operated." An adjustable damper which controls the acoustical filter on the pressure-operated portion permits either a unidirectional, bidirectional or combination non-directional pick-up pattern.

Ribbon microphones can be either velocity- or pressure-operated. In the RCA 77 microphone, pictured in Fig. 10 and shown in cross section in Fig. 11, both types are combined. The lower half of the ribbon is velocity-operated—that is, it responds to the *velocity* of the sound wave, because both sides of the ribbon are exposed to the air—and its output therefore decreases at low frequencies. This effect is sometimes desirable, especially for voice pickups.

Since it is open on both sides, this portion of the microphone is bidirectional. The output is a maximum when the sound source is facing either the front or the back of the "mike" and a minimum when the source is at either side.

The upper half of the microphone ribbon in Fig. 11 is pressure-operated. The felt labyrinth and the pipe enclosing the back of the pressure ribbon act as an acoustical shield that keeps the diaphragm velocity constant for a fixed sound pressure (sound intensity) over an extended range of frequencies. This section of the ribbon, therefore, has a better low-frequency response than the velocity-operated section. Since the sound acts only from the front of the

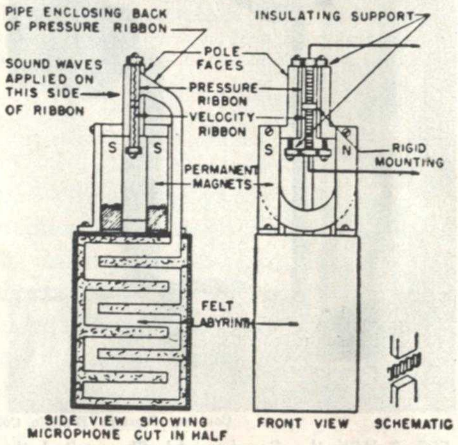


FIG. 11. A cross-sectional view of a combination velocity and pressure-operated ribbon microphone. An example of this type of microphone is the RCA 77 in which the size of the pipe back of the pressure ribbon is adjustable to produce various combinations of pressure and velocity-operated ribbon microphone characteristics.

ribbon in this case, this section of the ribbon is unidirectional.

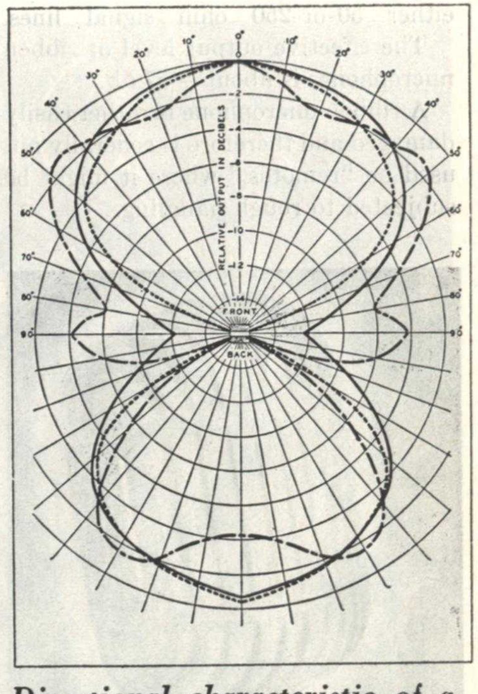
The directivity of this type of microphone can be varied by varying the size of the pipe to the acoustical filter. In the RCA 77 shown in Fig. 11, a

Courtesy RCA

FIG. 12. The RCA 74-B "all velocity" ribbon microphone which is widely used in studio broadcast pickups.

damper in the pipe can be adjusted to give a bidirectional, a unidirectional, or a non-directional pattern. The last is a combination of the first two patterns.

An example of an all velocity microphone is the RCA 74-B shown in Fig. 12. Its directivity is shown in Fig. 13 and its response curve in Fig. 14. Notice the decreased response at low fre-



Directional characteristic of a typical 74-B Junior Velocity Microphone.

——— 1000 cps
- - - - - 300 cps
—·—·— 8000 cps

FIG. 13. The directivity responses curve of the RCA 74-B microphone.

quencies. The variations in the response curve are caused by mechanical resonances in the microphone itself and are present in varying degree in most mi-

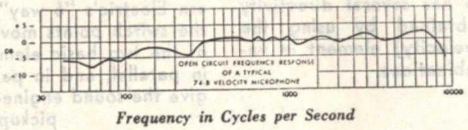


FIG. 14. Frequency response of the RCA 74-B microphone.

crophones of this type.

Since the ribbon resistance is quite low, generally only a fraction of an ohm, a transformer is used to couple to either 50-or-250 ohm signal lines.

The effective output level of ribbon microphones is about —57 db.

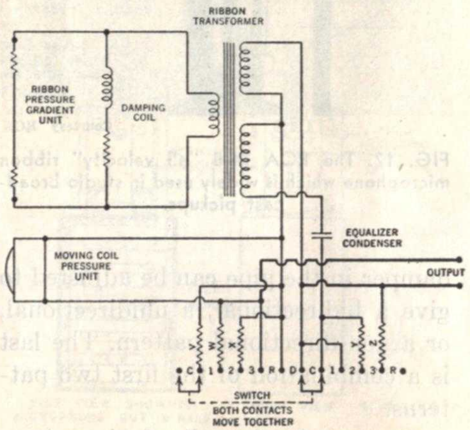
A ribbon microphone is rather easily damaged and therefore is generally not used on "remotes," where it might be subjected to rough handling.

Courtesy Western Electric Co.

FIG. 15. A view of the Western Electric 639 (Cardioid) microphone mounted on a table stand. This microphone has several directivity patterns which are obtained by using the dynamic and ribbon (velocity) element in various combinations.

Combination Dynamic and Ribbon Microphone. The Western Electric 639 microphone, an example of a combination dynamic and ribbon microphone, is shown in Fig. 15.

This microphone is designed to produce a wide variety of directivity patterns and is, therefore, very versatile. Fig. 16 is a simplified schematic diagram of the directivity switching circuits used in this microphone. The moving-coil pressure unit is the "eight ball" dynamic microphone that was shown in Fig. 9. The pressure gradient (velocity) unit is a special ribbon microphone. The resistors V, W, X, Y, and Z are used to change the relative level of the two units on the numbered positions of the switch which (as shown in Fig. 17) is on the back of this microphone. It permits switching in the bidirectional velocity ribbon (R), the non-directional dynamic element (D), or a combination of the two. The combination has a "cardioid" (heart-shaped) directivity pattern (C). In

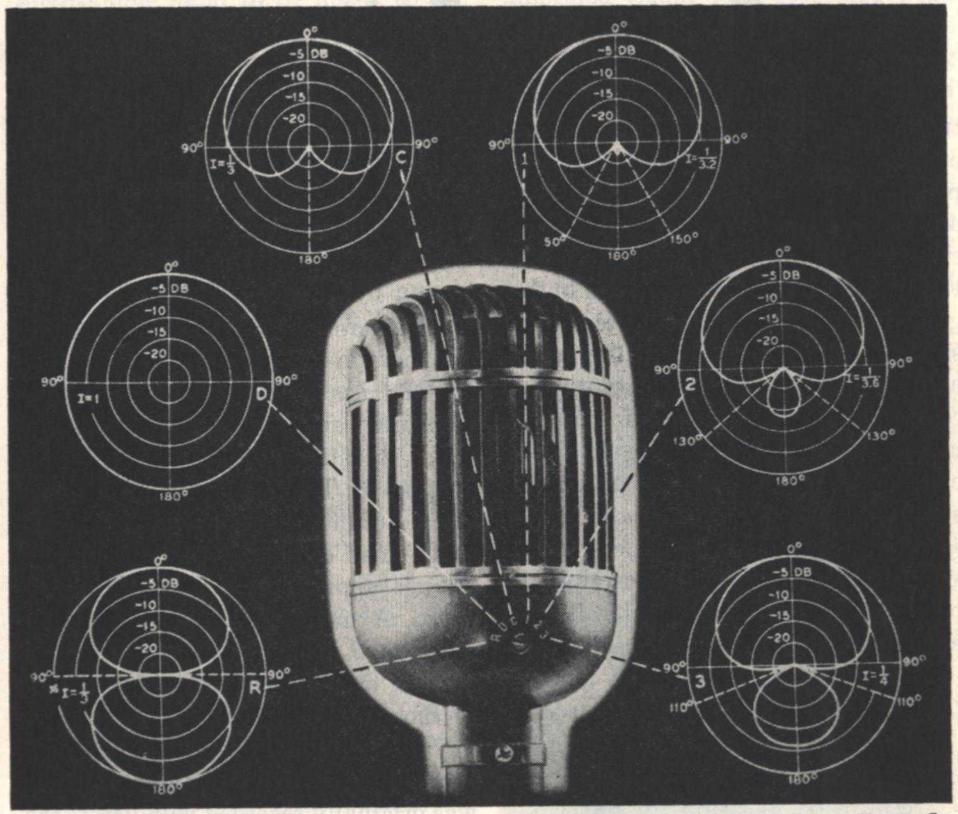


Courtesy Western Electric Co.

FIG. 16. Simplified schematic diagram of Western Electric's "6 way" cardioid microphone. As the switch points move from left to right, each of the two basic elements function individually, in parallel, and in parallel with attenuation, to give the sound engineer a choice of six different pickup patterns.

addition, three other switch positions (1, 2, 3) permit these two elements to be connected at different levels to produce combination directivity patterns that each have two dead spots.

This combination of two micro-



Courtesy Western Electric Co.

FIG. 17. By rotating the small flush switch in the rear of the Western Electric 639, six distinct patterns of sensitivity are successively brought into action. These patterns are (clockwise) bidirectional or velocity type, non-directional, cardioid or unidirectional, and three degrees of "hypercardioid" sensitivity. In the three latter positions, two areas of deadness, useful in shutting out unwanted sounds, exist at the rear of the mike.

phones produces a cardioid pattern over a wide range of frequencies. The various patterns possible make this microphone very useful in remote pickups, where a wide variety of acoustical characteristics may be encountered.

Transcription Techniques

Records and transcriptions constitute a major part of many broadcasters' scheduling. "Records" are the popular home-use 10- or 12-inch laterally cut records that operate at 78 r.p.m. and are played "outside-in"—that is, the pickup arm is placed in the groove on the outside of the record to start and the arm works to the inside as the record plays. "Transcriptions," recordings made especially for broadcast use, are usually 16 inches in diameter, and generally use a turntable speed of 33-1/3 r.p.m. to enable a full 15-minute show to be recorded on one side of the disc. These transcriptions may be either vertically or laterally cut and either "outside-in" or "inside-out."

As you learned in an earlier Lesson, two phonograph turntables are used in

Courtesy RCA

FIG. 18. An exterior view of the RCA type 70 transcription turntable used in broadcast programming. This unit operates at either 33⅓ or 78.26 rpm, plays either vertically or laterally cut records and transcriptions, can accommodate records up to and including 16", and has 6 filter positions for altering the frequency response of records and transcriptions.

most broadcast stations. These turntables can be located in one of the studios or in the announcer's booth, in which case they are operated by the announcers or by special personnel, or they may be in the control room and be run by the control room operator.

An exterior view of the RCA type 70 phonograph turntable is shown in Fig. 18, and an interior view in Fig. 19. Constancy of speed of the turntable is a most essential requirement in such equipment. The regulation of the driving motor must be such that variations of the load caused by the pickup arm do not appreciably affect the motor speed. A motor with poor regulation will have a tendency to produce "wows" on sustained notes, particularly those of low frequency. Naturally, anything that makes the disc turn at a speed different from that at which it was originally recorded will distort the sound in some fashion. Driving power for the turntable shown in Fig. 19 is furnished by a self-starting, two-speed, synchronous motor whose developed torque is large compared to the load variations, making the latter of small consequence. The flywheel on the main shaft further stabilizes the speed.

The turntable pickup head has two sets of coils, one for vertical and the other for lateral records and transcriptions. A 6-position switch on top of the cabinet allows either set to be switched in, and also allows various filters to be switched in. These filters are used to compensate for variations in the frequency response of the various brands of records and transcriptions used. They are also used to correct for the condition of the record; old records,

for example, tend to be noisy, so the high frequencies are deliberately attenuated by switching in low-pass filters when such records are played.

The output level of a turntable depends on the record being played, since the levels of records are not always the same. The output of this turntable is about —65 db when an average record is played.

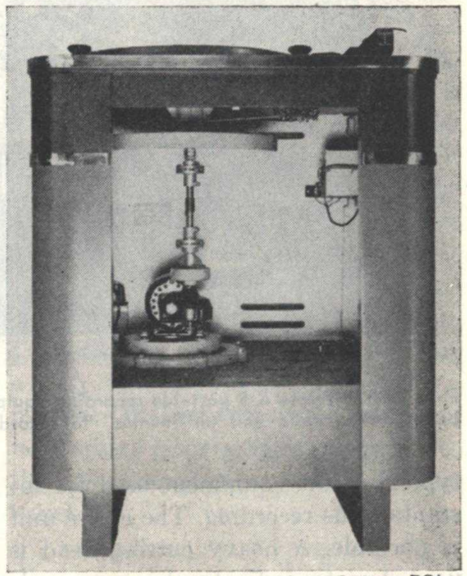
The output impedance of this unit is 250 ohms.

In operating a turntable, it is necessary to be sure that the pickup selector switch is on the proper setting for the pickup arm used (vertical or lateral), that the turntable speed switch is on the correct speed adjustment for the particular recording or transcription used, that the pickup arm is in the correct position (the record may be "outside in" or "inside-out"), and that the disc has been properly "cued." The last means that the pickup arm is at the spot on the groove where the announcements or music begins, so that no time is lost in waiting for the arm to reach that point after the record is started. This is usually accomplished by listening with headphones on an auxiliary amplifier so that each disc may be "cued in" before being put on the air.

INSTANTANEOUS RECORDINGS

The need often arises in a broadcast station for the use of instantaneous recordings. For example, when a network or other program occurs at a time when it cannot be broadcast, a record of it is made at the time and broadcast later.

Such recordings are made on a disc having an aluminum or glass base, covered with a cellulose nitrate lacquer. The recording is done by feeding the amplified signal to a recording head that translates the electrical signal into mechanical motion of a recording needle held in the head. The head and needle are moved over the disk at a regular rate by a driving assembly; this regular motion causes the needle to cut a spiral groove in the surface of the disc. At the same time, the motion of the needle caused by the input signal



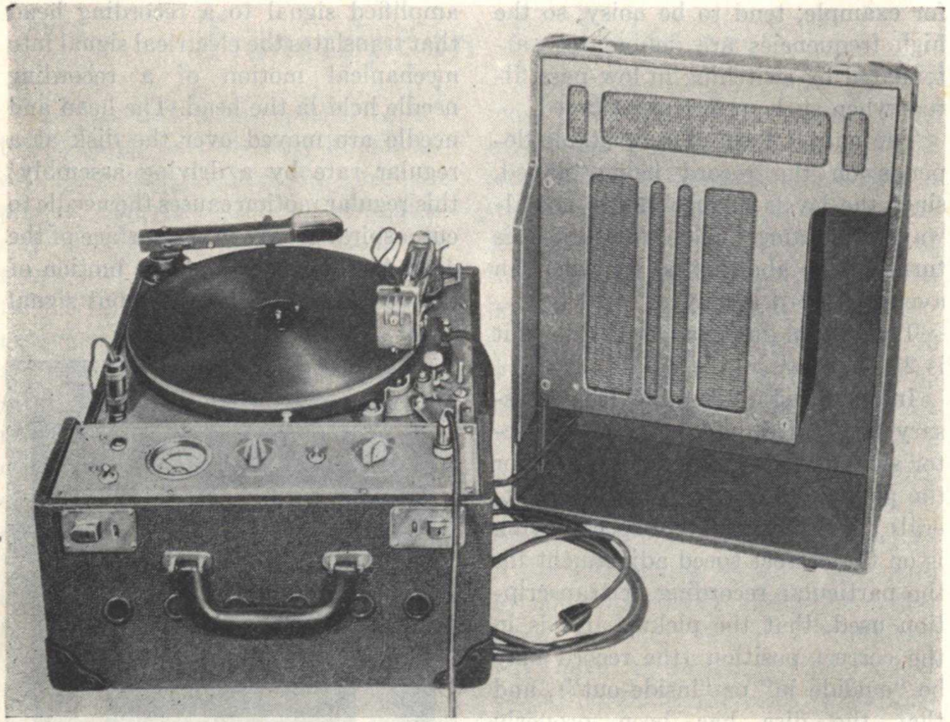
Courtesy RCA

FIG. 19. Interior view of the RCA 70 transcription turntable. A high torque motor and a heavy flywheel are used to assist in maintaining a constant speed of rotation despite variations in the load.

forces the needle to cut wiggles in the disc that represent the signal.

The lacquer with which the disc is surfaced is soft, but hardens somewhat after being cut by the recording needle. The record may be played back without further hardening, a fact that justifies the name "instantaneous recording." Such records may be cut at either 33-1/3 r.p.m. or 78 r.p.m. and are nearly always laterally cut.

The Presto K-8 shown in Fig. 20 is



Courtesy Presto Recording Corp.

FIG. 20. The Presto K-8 portable recording equipment shown here is typical of the equipment used both in the studio and on "nemos" to record programs for auditions and delayed programs.

typical of the equipment used for instantaneous recording. The entire unit is portable. A heavy cutting head is used. A meter indicates the proper volume level for recording: low volume results in under-cutting and a high noise level; high volume may result in over-cutting, in which one groove cuts into an adjacent groove.

After a record has been cut, the light pickup arm can be used to play it back through the amplifier and loudspeaker mounted in the top of the case.

Different makes of recorders have a number of refinements in the way of adjustments that are explained in the operating instructions for each type. The operator must become familiar with the particular features of the type for recording equipment being used and the methods of using it before he can produce a good "cutting."

Remote Broadcasting

A remote pick-up (known as a "nemo") is a program originating at a point not in the main broadcasting studios. A nemo may call for a complex set-up requiring a great number of microphones, or it may mean a broadcast from a newspaper office requiring only one microphone and no control operator at all.

Remote control equipment must have the same ability to mix and amplify the outputs of high quality microphones as has the main studio control equipment. The equipment must, however, be conveniently portable and therefore small in size and light in weight.

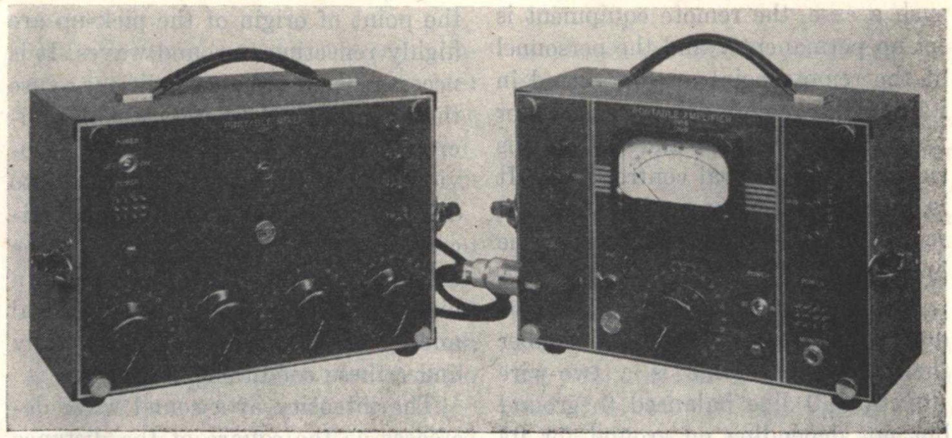
EQUIPMENT FOR REMOTES

A representative remote mixer-preamplifier, the RCA OP6/OP7, is shown in Fig. 21.

This remote control equipment consists of two units, the OP-7 portable mixer shown on the left and the OP-6 portable amplifier shown at the right.

The portable mixer unit has provision for four microphone inputs (the plugs are on the rear of the cabinet) and has a mixer for each input. Each channel has a separate amplifier stage using a 1620 tube (low-noise, non-microphonic 6J7). Both the input and output circuits may be either 30 or 250 ohms. The maximum gain of this unit is 8 db. The output level, of course, depends on the level of the microphone used. The usual level is —55 db, the maximum —24 db. The noise at this maximum output is 63 db below the output.

This unit is normally powered from a self-contained a.c. power supply using a 6X5 tube. However, for emergency use, the OP-7 as well as the OP-6 can be operated from batteries plugged into the power receptacle on the front of the panel.



Courtesy RCA

FIG. 21. There are many types of remote "nemo" equipment. The RCA OP-7 mixer, at the left, has separate gain controls for four channels, the OP-6 pre-amplifier at the right, can operate either directly from one microphone or as shown from the OP-7. Either of the portable units can operate from a.c. lines or from batteries which are carried in a separate box.

The OP-6 portable pick-up amplifier has three resistance-coupled stages using 1620 tubes to provide a 90 db gain to operate either from a low-level mixer such as the OP-7 or directly from a microphone. The normal output level of this unit is 8 db, although it is capable of producing a 19 db output.

The input impedance is either 30 or 250 ohms, the output impedance is 150 or 600 ohms. (Many telephone lines used for remote work have 600-ohm impedance.) A switch on the front panel permits using either the input from a line or the input from a standard microphone receptacle.

A step type of volume control is used. A VU meter may be used for visual monitoring of the output level, and headphones may be plugged into the monitor jacks to provide an aural check.

Simplex Remote Circuit. When there are many remotes that originate quite frequently from the same point (for example, an hourly news broadcast from a newspaper's offices), it may not be desired or necessary to send a remote operator for each broadcast. In such a case, the remote equipment is set up permanently and the personnel at the remote point are instructed in its use. The remote amplifier is set for normal amplification and the gain is ridden at the central control room. It is, therefore, necessary only to turn the remote amplifier on and off for the broadcasts.

As a matter of fact, even this can be done from the control room. Remember that the remote line is a two-wire ("metallic") line balanced to ground but not depending on ground for its operation. A d.c. voltage can be used, as shown in Fig. 22, to control the a.c. power to the amplifier remotely by means of a simplex circuit. One side of the battery B is grounded and the other side is applied through S_1 to the center tap of the secondary of T_1. Thus, when the patch cord is connected, and S_1 is closed, the d.c. is applied to both lines. At the remote position, the d.c. is taken from the center tap on the primary of T_2 through relay R. When R is energized, S_2 will close, thus turning the remote amplifier on.

This method will not work when there are repeating coils or amplifiers in the line that break the metallic circuits and prevent the d.c. path from being completed.

OPERATING REMOTES

The remote operator has to contend with many unfavorable conditions that do not exist in a regular studio designed for broadcast purposes.

Acoustics. Acoustical conditions may be such that the type of microphone and the method of placement play an important part in determining the success of a remote broadcast. The remote operator often finds, for example, that the surfaces of the walls at the point of origin of the pick-up are highly reflecting to sound waves. It is necessary under these conditions to use the directional characteristics of a microphone to its best advantage. Obviously, the operator would run into difficulty if he used a bidirectional microphone with one live side toward the sound source and the other toward a highly reflecting wall. A unidirectional microphone can be used successfully under these conditions, however.

The intensity of a sound wave decreases as the square of the distance between the sound origin and the point of pick-up. By increasing the distance between the sound source and a highly reflecting surface, it is possible to decrease the amount of reflected sound wave energy returned to the microphone. Also, by experimenting with the distance between the sound source and the microphone, any desired relationship between the original sound and reflected sound can be achieved. Decreasing the distance between the mike and the sound source gives a greater proportion of original sound at the mike, increasing the distance gives a greater proportion of reflected sound. Thus, the proper choice and placement of microphones play a very important role in remote pick-ups.

The time lag between original and reflected sound governs the apparent reverberation time (the time required for a sound to decrease to one-millionth of its original intensity) of the studio or hall. If the reverberation time is too great (large amount of reflected sound present at the microphone) the sound will be hollow and cavern-like. If the reverberation time is too small, as in heavily draped rooms, the sound will be flat and dull.

Wind sometimes causes a rumbling sound in a microphone used on an out-of-doors pick-up. For this reason, a shield is often used around three sides of the mike. A directional mike is nearly always used so that the dead side will contribute no unnecessary noise output to the amplifier. Dynamic microphones are usually favored for outside pick-ups where wind is a factor.

Remote Lines. Telephone lines used to broadcast remote programs are arranged for by the Chief Engineer or the Program Department as a rule. The telephone company is notified as to where the line is to be installed; it then installs a line for the broadcast (if there are enough broadcasts from a particular point, a line is permanently installed for this use) and "checks in" on the line to make sure it is continuous from the remote point to the control room. All lines to the telephone company offices that terminate on the nemo

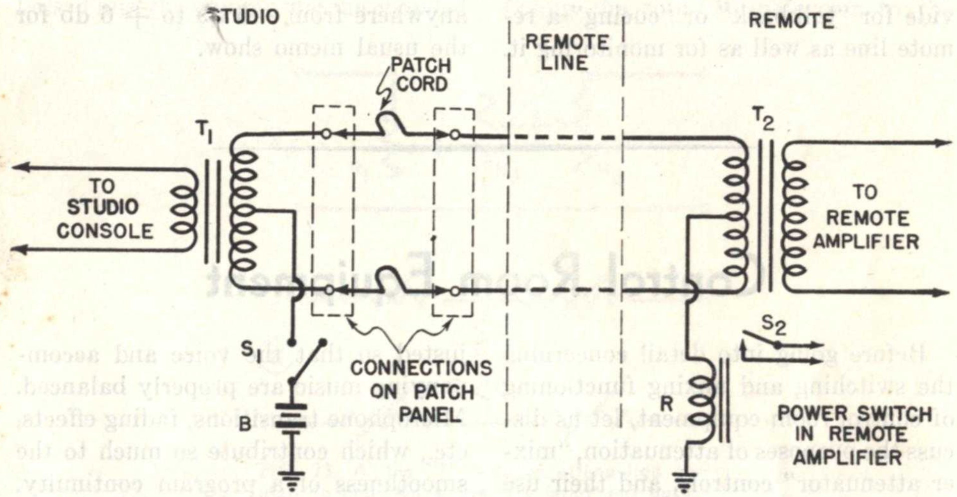


FIG. 22. How a "simplex" circuit can be used to turn the amplifier at a remote pick-up point on and off at the studio. The d.c. from B is applied to the line and relay R, which in turn operates the a.c power switch S_2 through the ground path. The audio signal operates through the "metallic" circuit that is, both lines, while the d.c. circuit operates "through ground."

portion of the patch panel are numbered for ease in reference. For example, the telephone company may notify the control room that the line installed to a certain remote point is on "pair 1128." The control room operator records this number as being the line used to that point, and talks to the telephone company test man over this line by whatever facilities are available. Most control room consoles provide for "talkback" or "cueing" a remote line as well as for monitoring it.

Regular telephone equipment can also be connected to the line.

Line Level. The minimum signal used with a remote line will vary. On a long line, for example, it may be necessary to use a minimum signal of +6 db to override line noise. (This is possible, of course, only if the +6 db will not cause excessive cross talk to adjacent telephone lines.) The zero reference at the remote point may be anywhere from —6 db to + 6 db for the usual memo show.

Control Room Equipment

Before going into detail concerning the switching and mixing functioning of control room equipment, let us discuss the purposes of attenuation, "mixer attenuator" controls, and their use in practical console circuits.

MIXING CIRCUITS

Volume controls for each pick-up as used on control consoles are marked "Mixer-1 Attenuator," "Mixer-2 Attenuator," "Mixer-3 Attenuator," etc., designating the number of the microphone or turntable with which the mixer attenuator is associated. A mixing circuit consists of a number of these controls arranged so that various program elements from separate channels can be combined in the proper proportions into one program. It may be necessary, for example, to use a microphone for a soloist, another microphone for an orchestra, a third for an accompanying choir, etc. The volume from each of these microphones must be adjusted so that the voice and accompanying music are properly balanced. Microphone transitions, fading effects, etc., which contribute so much to the smoothness of a program continuity, are produced by the control engineer at the mixing panel.

Since the volume controls in a mixer circuit are tied together, ordinary potentiometers cannot be used. If they were, a change in the resistance of one circuit would affect the over-all impedance and seriously alter the frequency response of the entire circuit.

To prevent this, fader controls of the "constant resistance" type were developed. These volume controls are of two main types, called T pads and H pads. Both are illustrated in Fig. 23.

You have studied the T pad in an earlier Lesson, but let's review its operation briefly now. It consists of three variable resistances, ganged on a common shaft and operated by a single control knob. As the control knob is

rotated clockwise, the values of R_1 and R_2 decrease; at the same time, the value of R_3 increases by an amount that maintains the total impedance, looking into or out of the pad, at a constant value. However, as the control knob is turned in this direction, the voltage drop across R_1 and R_2 decreases, and the drop across R_3 increases, so that more of the input voltage appears across the output terminals. Thus, the volume may be changed

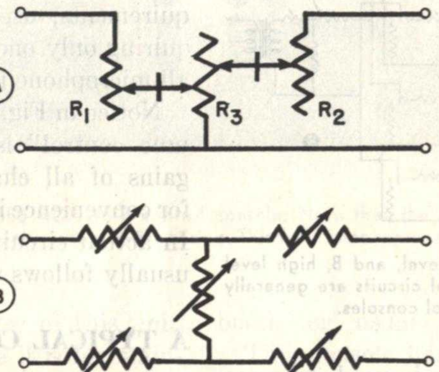


FIG. 23. A, the "T" type of attenuation pad used to maintain a constant input and output impedance for all values of attenuation. B, the balanced line equivalent, the "H" pad.

without altering the input or output impedances.

Generally, R_1, R_2, and R_3 are not continuously variable, but consist of step resistors with the level controlled in 1.5 or 2 db steps by switch positions on the mixer.

The T pad is used in unbalanced circuits where one side may be grounded, whereas the H pad is used in the balanced lines and circuits commonly used in broadcast work. (The H pad, as you learned in an earlier Lesson, consists of two T pads, connected as shown in Fig. 23.)

These mixers may be arranged in series, parallel, or series-parallel to match any impedance desired. In many installations, a special impedance-matching transformer is used after the mixing system to match the attenuator circuit to the load. This matching transformer may be used for isolating one circuit from another, thereby making it possible, say, for the primary winding facing the mixer circuit to be center-tapped to ground (balanced to ground) and for the secondary winding facing the input transformer to the amplifier to be grounded at one end (unbalanced to ground). Balanced circuits are usually used in mixing systems to eliminate cross-talk.

Mixing Level. When the mixer pads are placed in the circuit immediately following the microphones, we have what is known as a "low-level" mixer circuit, meaning that the signal level at the pads is extremely low before amplification. When the attenuators are placed in the circuit immediately following the pre-amplifiers, we have a "high-level" mixer system. Fig. 24 illustrates these two types of mixer systems. Notice that a separate preamplifier must be used for each microphone

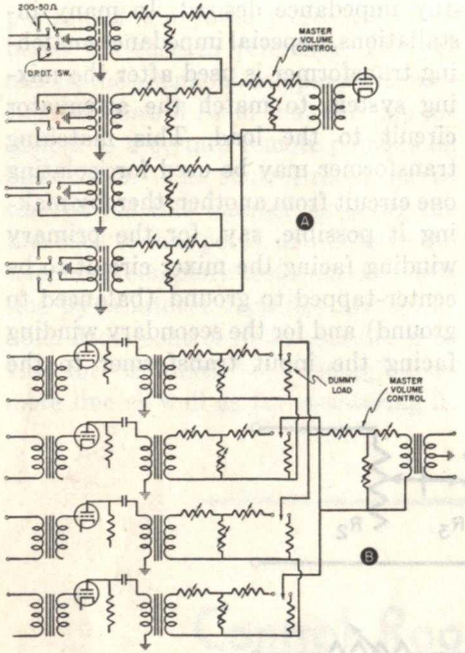


FIG. 24. A, typical low level, and B, high level mixing circuits. High level circuits are generally used in control consoles.

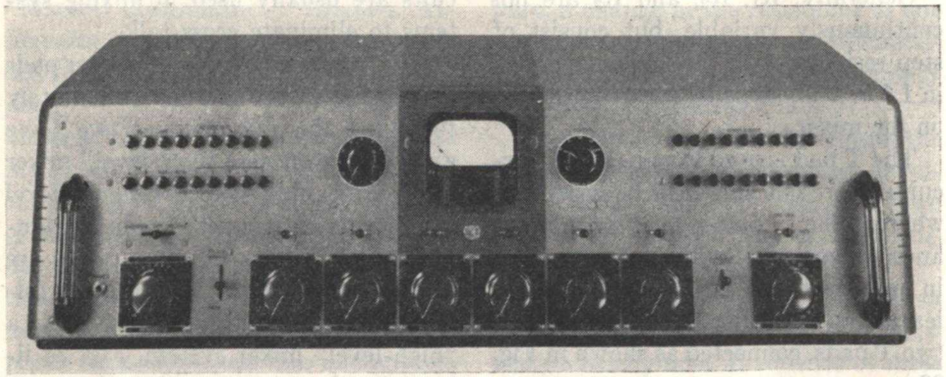
if high-level mixing is used, whereas the low-level mixing circuit permits the use of only one preamplifier for the multiple channel input.

High-level mixing gives a greater signal-to-noise ratio than does low-level, because of the fact that, in the former, the signal is amplified before being passed into the mixing and switching system (which is often a source of noise). High-level mixing is also more dependable, since failure of a single-preamplifier will kill all microphone inputs in a low-level system, but only one microphone input in a high-level circuit. Nearly all control room installations use high-level mixing. Remote amplifiers, where space is limited by weight and portability requirements, use low-level mixers requiring only one preamplifier tube for all microphone inputs.

Notice in Fig. 24 that a "master volume control" is used to control the gains of all channels simultaneously for convenience in over-all gain control. In actual circuits, this master control usually follows the program amplifier.

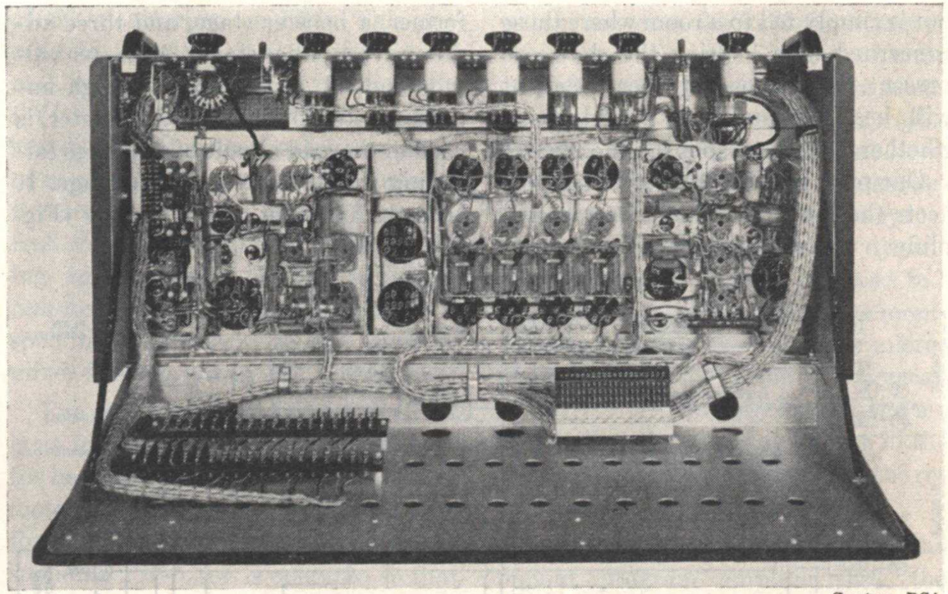
A TYPICAL CONTROL CONSOLE

Fig. 25 shows the panel of the RCA 76-B consolette. This equipment may be used to handle two studios from a single control room in a small station or to handle a single studio in larger stations. A bottom view of this unit, showing how accessible it is for maintenance and repair, is given in Fig. 26. Fig. 27 is a simplified schematic of the control circuits.



Courtesy RCA

FIG. 25. Front view of the RCA 76-B control console described in the text. This console is used as master control in smaller broadcast stations, and as an individual studio control in larger stations. The four rows of interlocked buttons facilitate operation and eliminate operator error.



Courtesy RCA

FIG. 26. A bottom view of the RCA 76-B control console. Note that the top of the console folds back to permit easy access to the wiring. This facilitates repair of this unit.

A total of six 250-ohm mixer controls are provided in this equipment: 1 and 2 control the two microphones in Studio A and 3 and 4 those of Studio B. Controls 5 and 6 can be used with two sets of push keys to select any two of six remote lines or two transcription inputs.

Ahead of the four microphone mixers are four preamplifier stages that amplify the outputs of the studio microphones before they are fed to the mixer system, thus producing high-level mixing. A switch K-7 on the input of the fourth preamplifier allows it to be switched to a fifth microphone, which may be used as an announcing microphone in the control room or at some other point, such as a transcription booth or usual announcers booth.

This console has four sets of push keys, PK-1, PK-2, PK-4, and PK-5. The mixer-5 input, PK-1, permits turntable TT-1 or TT-2 to be connected directly to the program bus or any of the 6 remote lines to be connected to the console through T_9. The mixer-6 input PK-2 permits any other one of these 8 inputs to be connected to the console. Each set of push keys consists of nine buttons interlocked so that only one button can be used at a time (the ninth is an "off" switch). When any program-audition switch (K-1, K-2, K-3, K-4, K-5 or K-6) is thrown to the program (PGM) position, the output of the corresponding mixer is placed on the program bus. Likewise, when the switch is thrown to the audition (AUD) position, the signal is placed on the audition bus. (An audition is a show originating in a studio or at a remote point that is not put on the air, but is simply fed to a room where those concerned with hearing the show are present. Sometimes a prospective client will hear an audition to determine whether or not to sponsor the show.)

One push key (MON) of PK-4 connects the input of the monitoring amplifier, through proper bridging resistors, across the output of the program amplifier for monitoring the program being broadcast. Another push key on PK-4 (AUD) connects the monitor amplifier to the audition bus, permitting the carrying on of an audition in one studio while another studio or remote line is on the air. A phone jack on the output of the program amplifier permits headphone monitoring of the program.

The program bus is connected to the input of the program amplifier. This amplifier consists of an input transformer, a booster stage, and three additional amplifier stages. A master volume control, consisting of a high-impedance step-by-step potentiometer, is used in the grid circuit of the stage following the booster amplifier stage. It is at the far right of the console (Fig. 25).

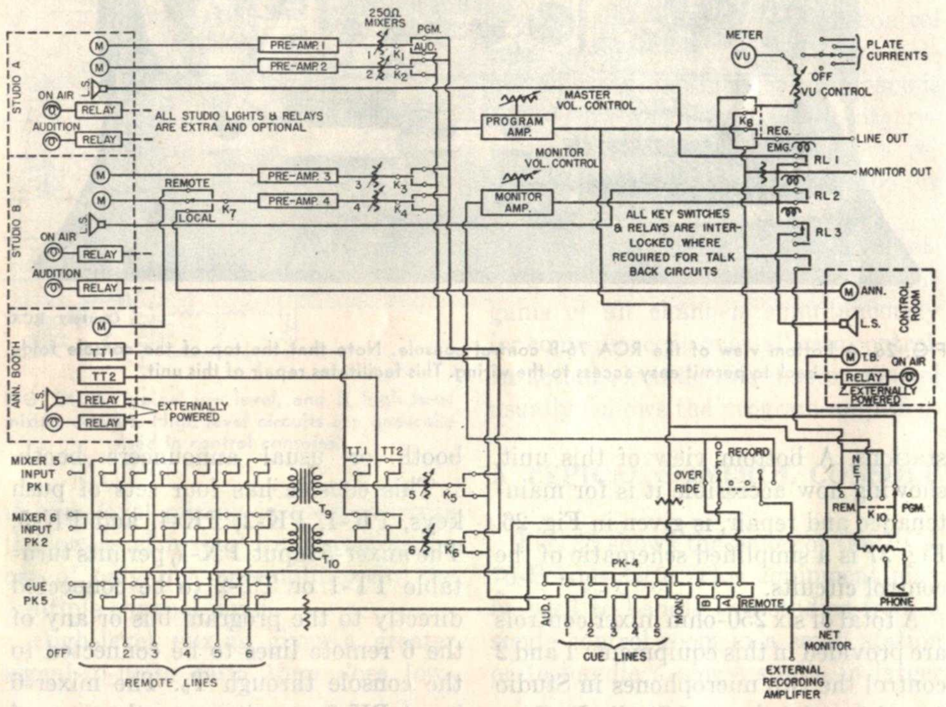


FIG. 27. The basic schematic of the RCA 76-B studio control console.

Two push keys (A and B on PK-4) are used for talk-back from the control room to a studio, one for Studio A and one for Studio B. Pressing the button of either key connects the talk-back microphone to the input of the monitor amplifier that feeds the studio speakers when the talk-back microphone is in use.

Six members of the fourth set of push keys (PK-5) connect to the incoming remote lines. These six keys are also normally connected through the OVERRIDE switch to the output of the monitoring amplifier. This permits an operator on a remote line to call in on the line at any time the OVERRIDE switch is on. He will be heard on the monitor speaker, although, of course, he will not be "on the air." Pressing any one of the six keys will feed the signal output of the monitoring amplifier into the corresponding remote line, provided that the corresponding push key in the input of mixer 5 or 6 is not in use.

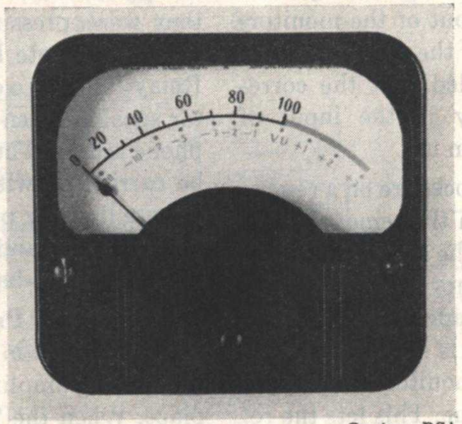
The operating procedure on a remote is as follows: When the remote operator has called into the studio on the remote line (this occurs before it is time for the remote broadcast), the corresponding PK-5 key is operated so that the output of the monitor amplifier is fed to the remote line. This lets the remote operator hear the program in progress at the studio, so that he will be able to hear the cue for the start of his program at the remote point. (That is why the PK-5 keys are called "cue keys.") When this pre-arranged cue, which often takes the form of a studio announcement, has occurred, the studio operator presses the proper key in the mixer-5 or mixer-6 input (PK-1 and PK-2) to connect the remote line to the input of the amplifier that feeds the program amplifier. Since the cue keys and the mixer-5 and mixer-6 keys are interlocked, this action releases the cue key. Thus, only one line is necessary for any remote control program; either the cue or the program can be fed over the same line by pressing the proper keys.

The PHONE jack which through the remote (REM) position of switch key K-10 can be connected to any of the six remote cue keys is used to listen to any one of these lines. The control operator can talk to the remote line by pressing the REMOTE key of PK-4 which connects the talk back microphone to the input of the monitoring amplifier, and the output of the monitoring amplifier is connected through a pad and RL-3 to the remote cue keys. (The talk back keys are not mechanically interlocked with the cue keys; if they were, pressing one would disconnect the remote line to be talked to.) Relay RL-3 also disconnects the monitor speaker when the talk back microphone is used. Thus, a conversation can be carried on with any one of the six remote lines by plugging phones in the PHONE plug and holding down the remote-line talk-back key.

Emergency Provisions. Should the regular program amplifier fail, the monitoring amplifier can be used in its place. When the "Line-Out" (output) switch is thrown to the "Emergency" position, the outgoing line is connected through a bridging resistor to the output of the monitoring amplifier. Thus, to use the monitoring amplifier in place of the regular amplifier, the operator merely places the program signal on the audition bus and presses the audition button in PK-4 at the input of monitoring amplifier. Throwing the output switch to the emergency position also transfers the volume indicator to the monitor output so that the level can be properly adjusted.

An emergency B supply (not shown) is provided for the four pre-amplifiers. If the B power normally obtained from the program amplifier rectifier should fail, throwing a switch on the power supply wall box will allow the B power to be obtained from the monitoring amplifier rectifier. The power supply is in reality two separate supplies—one for the program amplifier (and preamplifiers) and one for the monitoring amplifier.

Light Relay Operation. Provision is made for supplying d.c. power (12 volts) to four studio signal light relays. The four lights are: (1) Studio A "On-Air," (2) Studio A "On-Audition," (3) Studio B "On-Air," and (4) Studio B "On-Audition." The "On-Air" light will come on in either studio whenever the "Program-Audition" key switch (K-1, K-2, K-3, or K-4) associated with that studio is thrown to the "Program" position and the "Line-Out" key switch (K-8) is in the "Regular" position. Operating either key to the "Off" position will turn out the light. Similarly, throwing the "Prog-Aud" key to the "Audition" position and closing the "Audition" key on PK-4 will turn on the "Audition" light of the corresponding studio.



Courtesy RCA

FIG. 28. A view of the standard VU meter used for visual monitoring of broadcast program signals. Generally the operator strives to maintain the program level at the zero VU (100 on top scale) mark.

THE STANDARD VU INDICATOR

As you have learned, the VU meter is an essential part of control room equipment, because it gives a visual indication of the program level.

A front view of a standard VU meter is shown in Fig. 28. The schematic diagram of Fig. 29 shows the circuit used to bridge the meter across program lines of individual studio output lines. The total impedance presented to the line is about 7500 ohms, of which 3900 ohms is in the meter and about 3600 ohms in an external resistor. The dynamic response of this meter is standardized: it must be such that, if a 1000-cycle voltage of sufficient amplitude to give a steady indication of 100 on the voltage scale or 0 db on the decibel scale is suddenly applied, the pointer will reach 99 in 0.3 second and overswing the 100 mark by not more than 1.5 per cent.

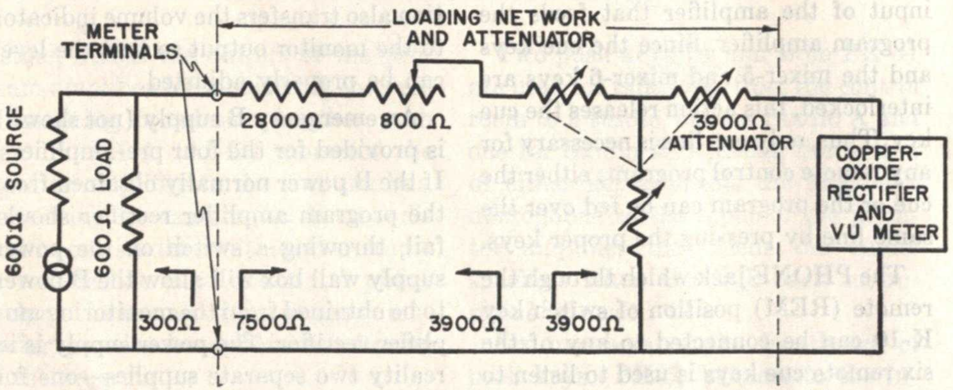


FIG. 29. How a standard VU meter is connected to a 600-ohm line. Note that the VU meter impedance is 7500 ohms, which is so high that it can be connected across a 600-ohm load without affecting the circuit. The attenuator shown can be used to allow line levels above 0 VU to indicate only 0 VU on the meter, that is, to shift the reference level.

PATCH-PANELS AND REMOTE LINE TERMINATION

As you have already learned in this and other Lessons, patch panels are widely used in the control room to aid the operator in the performance of his duties.

Fig. 30 shows a typical panel designed for use on an individual studio relay rack that houses the preamplifiers and line amplifier. As you can see from the various labels, the microphone outputs, preamplifier inputs and outputs, mixer inputs and outputs, etc., are terminated on jacks so that patch cords may be used to route a signal in any way desired should trouble occur in any particular circuit.

All incoming remote lines are usually terminated at jacks on a patch panel known as the "nemo" panel. These lines, when they are not in use, are often patched in to a "cue amplifier"—a small auxiliary amplifier and speaker. The control room can then be called directly at any time from the remote point. (Of course, when using the console just described, cue amplifiers are not necessary, for the remote operator can call in when K-10 is in the OVERRIDE position.)

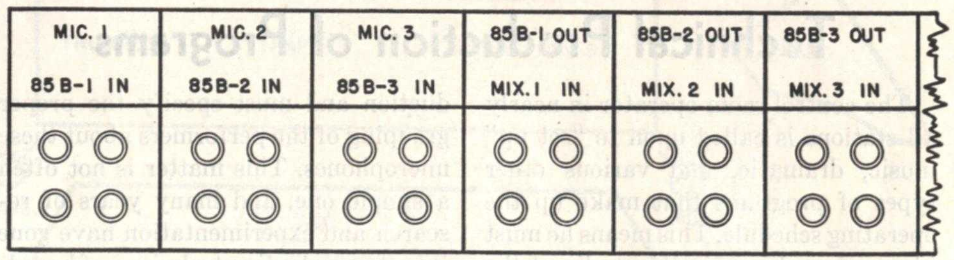


FIG. 30. Drawing of typical patch panel arrangement for a studio rack. Top line is associated with upper row of jacks and bottom designations are for bottom row of jacks. This particular portion of a patch panel permits access to the three microphone lines, the input and output of their three 85-B preamplifiers and the corresponding mixer inputs.

Sometimes there are two separate lines installed between a remote point and the nemo board, one being the regular broadcast line and the other a "talk line" used for direct communication between the remote operator and the main control room. In this case, a telephone with an alternator ringer (usually of the hand-crank variety) is installed on the talk line at the remote point, and the talk line is terminated on a "ring-down" in the main control room. (A ring-down is an indicator that rings a bell softly and lights a light when it is actuated.) When the remote operator wants to get the attention of the studio control room, he rings the ringer which in turn operates the ring-down at the studio and the control operator then talks to the remote point via the talk line.

► Each station, of course, differs in the equipment it uses. If you become a control operator, one of your first steps must be to acquaint yourself with the equipment in your station that you can use to correct quickly any trouble that may occur. In almost any station, you will find that it is seldom necessary to have to repair equipment during a program, since emergency switching and patching is provided to re-route the signal into other channels.

Technical Production of Programs

The control room operator in nearly all stations is called upon to "set up" music, dramatic, and various other types of programs that make up the operating schedule. This means he must place microphones in the studio at the proper points to fit the particular production and must specify the proper grouping of the performers about these microphones. This matter is not often a simple one, and many years of research and experimentation have gone into the production technique of broadcasting. A knowledge of the fundamentals of the art as they affect the technical operator is necessary to enable him to meet various situations in a capable manner.

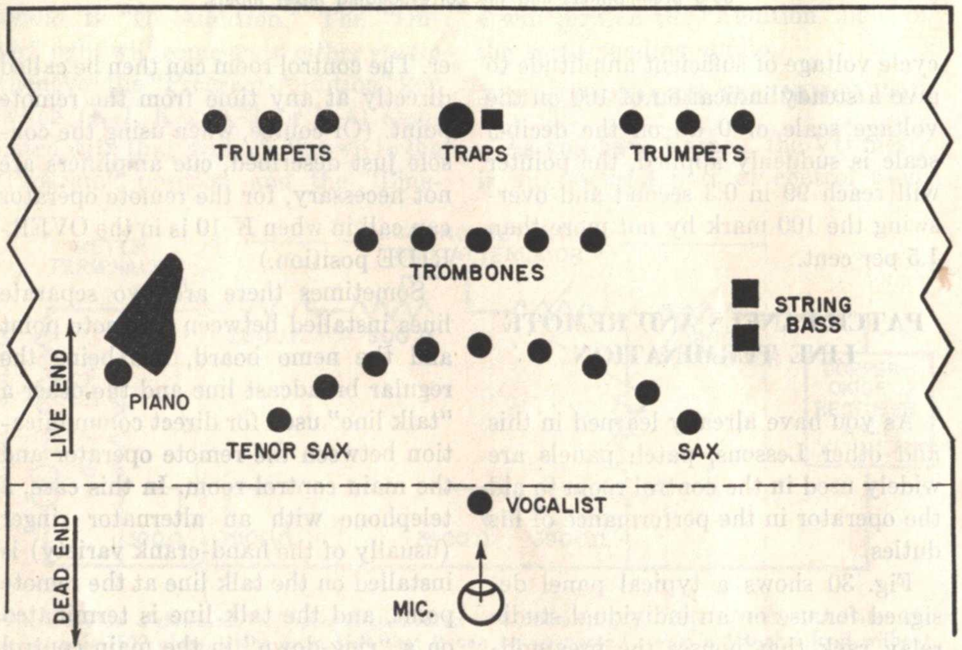


FIG. 31. One microphone can be used to pick up a properly placed orchestra when the broadcast originates in the "live" end of a "live-end, dead-end" studio.

Microphones are "spotted" in the studio to discriminate against unwanted sources of sound, and to obtain any desired relation between sounds from different sources. The operator, therefore, must be thoroughly familiar with the pick-up patterns of the types of microphones being used. In this regard, you should remember that the microphone directivity patterns generally shown assume a perfectly reflectionless room, and that the degree of "liveness" in a studio will substantially affect the pick-up pattern. If you become a control-room operator, learn what the actual pick-up patterns of your microphones are in your studios.

In a live studio that is properly designed, there is generally good sound diffusion and dispersion as well as substantial reinforcement of the overtones of musical instruments. For these reasons, one microphone is usually all that is necessary to pick up an entire orchestra whose instruments are properly grouped about the microphone. The instruments with the highest volume outputs, such as drums and brass instruments, are generally placed farther from the microphone than are such low-output instruments as strings and woodwinds. If a directional microphone is used, however, these louder instruments can be placed in a lower-sensitivity area of the microphone. A typical seating plan for an orchestra in a live-end, dead-end studio is shown in Fig. 31.

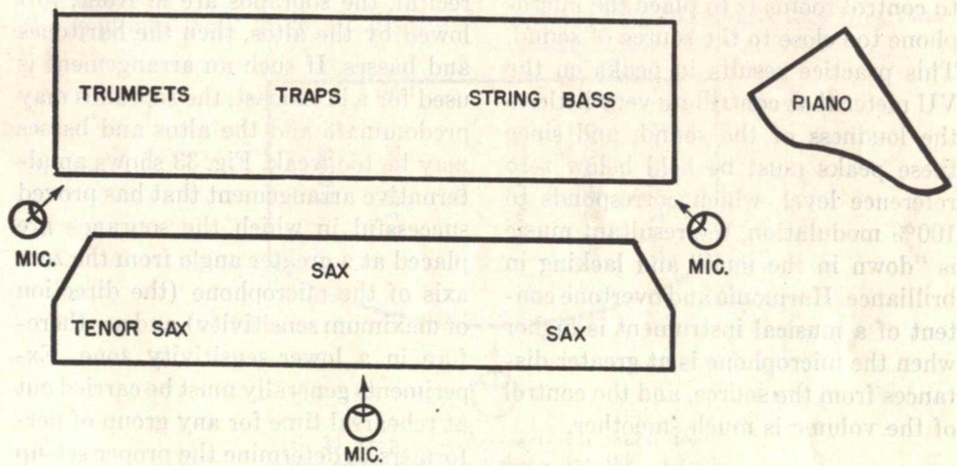


FIG. 32. How three microphones could be used in a "dead" studio to assure adequate pickup of a small orchestra.

If the studio is dead (that is, there is not sufficient sound reinforcement for naturalness), two or more microphones must be used to provide adequate pick-up; the use of only one microphone for a large group of instruments will result in a "thin" sound lacking "body." Fig. 32 shows how three microphones might be used in a dead studio for an orchestra pickup. This type of setup is naturally more complex and difficult to handle, since it requires the proper blending of the outputs of three different microphones.

In general, it is best to use the fewest microphones that will give adequate pick-up. If several must be used, each should be placed so that it is in a comparative null point in the pick-up patterns of the others. This will make it easier to control and blend the microphone outputs at the console.

Studio acoustics will also affect the optimum spacing to be used between microphone and musical instruments. The most common error of newcomers to control rooms is to place the microphone too close to the source of sound. This practice results in peaks on the VU meter that contribute very little to the loudness of the sound, and since these peaks must be held below zero reference level, which corresponds to 100% modulation, the resultant music is "down in the mud" and lacking in brilliance. Harmonic and overtone content of a musical instrument is higher when the microphone is at greater distances from the source, and the control of the volume is much smoother.

VOCALS AND CHORAL GROUPS

Vocalists and vocal groups present special problems to control room operators.

Vocalists. There are two general types of vocalists in broadcasting. The "crooner" type, who sings mostly from the upper larynx and throat muscles, has low volume and small dynamic range and thus must be placed close to the microphone. The "operatic" type, on the other hand, who uses his chest muscles when he sings, has a much greater dynamic range and must be placed a minimum of 3 feet, preferably 4 to 6 feet, from the microphone in even comparatively dead studios. A distance of 10 feet has been used in modern live studios without impairing the softer passages. It is worth the trouble to experiment, if possible, with the placement of a singer of this sort, because a much greater dynamic range and brilliancy of voice, together with smoother control, can be secured by finding and using the optimum microphone distance.

Vocal Groups. Proper balance is sometimes difficult to achieve when vocal groups such as choirs are to broadcast. Generally, in an auditorium recital, the sopranos are in front, followed by the altos, then the baritones and basses. If such an arrangement is used for a broadcast, the sopranos may predominate and the altos and basses may be too weak. Fig. 33 shows an alternative arrangement that has proved successful in which the sopranos are placed at a greater angle from the zero axis of the microphone (the direction of maximum sensitivity) and are therefore in a lower-sensitivity zone. Experiments generally must be carried out at rehearsal time for any group of performers to determine the proper set-up in a given studio.

PIANO AND ORGAN PICK-UP

Piano music, which is produced by a sudden striking of strings by mallets, has sound wave-fronts that are extremely sharp and therefore have high peak factors (the ratio of the peak volume to the effective sound volume is large). For this reason, piano music should be peaked about 2 VU lower than the general program level. A distance of close to 15 feet in a modern live studio, or 8 to 10 feet in a dead studio, should be used between the piano and the microphone to achieve brilliancy of piano reproduction.

Electric Organ Pickups. The electric organ uses an electric amplifier and a loudspeaker that contains a "reverberator" or "chopper" drum in the top of the cabinet from which the sound emanates. To obtain the maximum effect resulting from the design of this cabinet, the microphone should be suspended 6 to 8 feet above the top of the cabinet. A unidirectional microphone should be used to prevent any chance of bothersome reflections from the ceiling.

Pipe Organ Pick-Ups. Because of the comparatively large air pressures used in pipe organs, it is usually well to use a distance of approximately 25 feet or more between the pipe outlets and the microphone to obtain high harmonic content and smooth control.

DRAMATIC SHOWS

Usually in a dramatic show only one microphone is used for the entire cast, with a separate microphone for sound effects. As each actor speaks his lines, he steps up to the mike at a distance determined by level checks at rehearsal, sometimes approaching from the fading zone (the low-sensitivity areas at the back of unidirectional microphones) into the sensitive area to create the illusion of approaching the scene of action. He sometimes leaves in the same manner.

When "board fades" are marked on the script, the operator fades the entire studio set-up by turning down the master gain control. Rehearsals are very necessary for dramatic productions of this type.

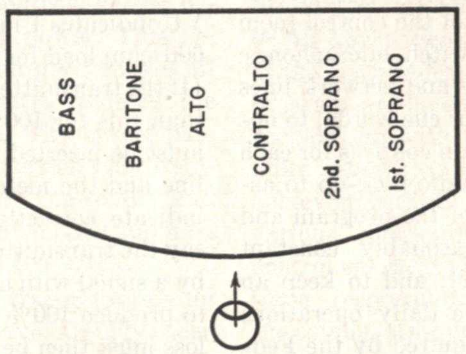


FIG. 33. Proper arrangement of a choral group when using one microphone for the pick-up. Note that the arrangement differs from the arrangement normally used.

REHEARSALS

The importance of rehearsals does not lie solely in the fact that they benefit the cast of a program. Just as important—perhaps more important—is the fact that they give the control room operator a chance to perfect the co-ordination of hand, ear, sight, and sound necessary to blend the audio components of a given performance properly without evidence of technical manipulation. Mixer adjustments or settings for the level control of each microphone may thus be determined in advance so that when the program is "on the air" the output level of each microphone and the over-all level will be correct.

In many broadcast stations, the studio is "pre-set" during rehearsals—that is, all microphones are turned on and the mixers are set at the proper level. Then, when the studio takes the air, only one button has to be pushed on the master control panel to connect that studio to the program amplifier.

If a regular production man is not assigned to a show, it is well for the operator to ask the orchestra conductor or show director to listen to the "balance" when the operator has placed the microphones to the best of his ability during rehearsal.

Control Room Operating Practice

The main duties of the control room operator are: to switch microphones, studios, and remote and network lines at the proper time or cue words; to operate the various gain controls for each microphone of a studio pick-up to assure proper blend of the program and to maintain a reasonably constant output volume level; and to keep an accurate log of the daily operations of the station as required by the Federal Communications Commission.

In addition, he should be a master of operating technique, which is the art of performing the necessary switching and mixing operations without making the listener aware of them.

Operating the gain controls properly is one of the operator's most important jobs, and the one in which the expert is most obviously superior to the beginner. Let's learn some practical pointers that will help you do this job well.

RIDING GAIN

One of the essential duties of the control room operator is to "ride gain"—that is, to make certain that the volume level seldom exceeds the value at which 100% modulation of the transmitter occurs. This point is generally marked zero VU on the VU meter. Zero VU indicates 1 milliwatt of power in a 600-ohm load for sine-wave excitation. (If the transmitter requires more power than this for 100% modulation, a loss must be inserted between the program line and the meter to make the meter indicate correctly. For example, let's say the transmitter must be modulated by a signal with a power equal to 4 VU to produce 100% modulation. A 4-VU loss must then be inserted between the line and the meter to make the 100% modulation mark on the meter indicate correctly.)

As you know, complex program waves are not sinusoidal, and therefore, when the program energy is sufficient to cause a deflection up to 0 VU, or 100 on the linear scale, instantaneous peaks having powers of several times 1 milliwatt may exist even though the average power may be only a fraction of 1 milliwatt. "Zero volume level," then, for monitoring program energy, is simply an arbitrary point, not to be thought of as measurable by any rigid electrical units of current, voltage, or power. For that matter, it often cannot be measured on even the VU meter, because, as you just learned, the meter is not accurate when the signal is not sinusoidal.

The operator riding gain must recognize certain basic facts about various types of programs to do his job properly. Symphony music, for example, will have many pianissimo passages that are extremely low in level, and may then suddenly run into a passage in which every instrument is contributing its utmost in the way of volume. A control engineer at the mixing panel can ruin a classical selection of this type by bringing all low passages up to a considerable indication on the volume indicator, then cranking down suddenly on the gain controls as the orchestra increases its power. This erroneous practice naturally removes all the color and dramatic value of the original composition. The lowest passages should be brought up only barely enough to "jiggle" the meter, and care should be taken not to suppress the loudest passages too much.

Recordings and transcriptions of symphony music have already been compressed in the recording process to a dynamic range suitable for broadcasting. This is done to avoid overcutting of the disc on the loudest passages and to bring the softer passages up above surface noise. With such records, it is usually necessary for the control operator only to set the level so that the peaks correspond to 0 VU or 100 on the scale and "let it ride."

When voice and music are to be combined, as when a vocalist sings with an orchestra, the two must be balanced by placing the microphones properly and by adjusting the gain controls of these microphones so that the voice stands out over the musical background. The operator must keep in mind the differences in peak factor of voice and music that we mentioned earlier, and also must remember that musical instruments such as the banjo and piano have extremely high peak factors and sharp wave-fronts, and should be peaked about 2 VU lower than the zero reference on the control meter to avoid overmodulation of the transmitter on peaks.

It is obvious that no set rules can be drawn up to guide a control man responsible for the proper blending of a musical show. In the final analysis, the ear is the best judge; when the faders are so adjusted that a pleasing combination of voice and music is heard, the control operator's job is well done.

► All key network stations and some of the large independent stations have program producers who work in close cooperation with the control room on mike set-ups and the balance to be achieved between the outputs of the various microphones. However, the control operator himself must shoulder these responsibilities in almost all the smaller stations.

The Human Ear. Since it is the intent and purpose of broadcasting to appeal to the listening interest of the radio audience, it is obvious, as we said earlier, that the human ear is the final judge of the excellence of operating technique.

The human ear is far from perfect in its response. For example, it was found that adding harmonics or overtones to a 40 db fundamental G tone (392 cycles) increased the intensity, according to a volume indicator, only to 40.9 db—but, according to listeners, the loudness level was raised 8 db.

It is clear, then, that volume-indicator measurements of intensity do not show what loudness sensation a sound will produce. The loudness sensation produced by a sound having a given meter reading will depend, among other factors, on the number of harmonics present and the phase relationships of these harmonics. This is why it often happens that two voices that are peaked the same amount on a volume indicator sound far different in loudness. If such voices are to sound equally loud, they must be peaked different amounts—sometimes as much as 3 to 6 db different.

Another reason for a difference between meter readings and loudness sensations is that the standard VU meter is an r.m.s. indicator rather than a peak indicator. Fig. 34 shows a complex voice wave of a shape that causes difficulties for this reason. Although this wave has high peaks, its r.m.s. value is low, and it gives, therefore, a low indication on a VU meter. Another wave having much lower peaks might give the same VU reading, but it would be far less loud. The only way to overcome this difficulty is for the control operator to monitor by ear as well as by eye.

The listening habits of the radio audience must also be taken into consideration by the operator at times, particularly when symphonic music is being played. The average symphony has a great many soft passages; consequently, a listener usually has the volume on his set turned up considerably more for symphonic music than he would have if he were listening to dance music, whose average intensity level is more uniform. Therefore, when announcements are made between symphony selections, it is well to peak these announcements 3 to 6 db lower on the VU indicator than the musical selections are peaked, both to avoid the impression of a "roaring" announcement that may seem very loud to the listener whose receiver is turned up high, and to avoid dwarfing the start of the next musical selection, which may be very low in volume.

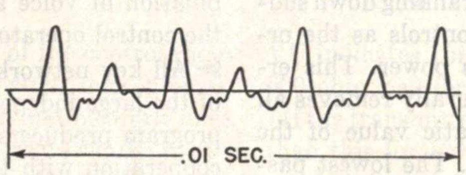


FIG. 34. A typical example of a complex voice waveform with high peak factor. This type of wave, has a low average power.

This is particularly important in "dead" studios, where the facts that there is no reinforcement of the musical overtones and that there is a great amount of high-frequency absorption reduce the apparent loudness of the musical performance. Since the announcer is usually within a few feet of the microphone, the lack of reverberation has little effect on the loudness of his speech. Therefore, it is necessary to peak the announcer's voice much lower in relation to the music to assure equal loudness sensation in a dead studio than in a live one.

KEEPING A LOG

One of the very important duties of a broadcast operator is keeping a log (the detailed record of station operation). The Federal Communications Commission requires that all broadcast stations maintain two logs: the program log (which we will study shortly) generally kept by the studio control operator, and the operating log

WIRE OPERATING SCHEDULE

MONDAY JUNE 18, 1945

STU	START	FINISH	CODE	TITLE OF PGM & DETAILS	ANNCR
ANN	6:29½			Sign On	NEHRLING
REM Remote	6:30 from Indpls Stockyards	6:33	REM-S	Livestock Reports (PSD)	"
ANN	6:33	6:59	ET-C*	Dawn Patrol (North Side Chev)	"
ANN	6:59	______	*	American Loan 100	
C	7:00		NBC-C*	WORLD NEWS ROUNDUP (STANDARD OIL)	"
ANN	7:14½	______	*	Nervine ET	"
ANN	7:15½	8:14	ET-C*	Musical Clock (Hooks) Gordon	"
ANN	8:14	______	*	Wonder Bread ET	"
ANN	8:15	8:29	LOC-C*	News Hilights (Strauss) Knox	"

FIG. 35. A portion of a sample broadcast operating schedule. This schedule is used as a guide for the operator in "setting up the shows" for the day. The program log as required by the FCC is prepared from the program actually "on the air" and, although quite similar to the operating schedule, may differ in some details.

maintained by the transmitter operator. Sometimes both these records are maintained by the transmitter operator.

Program Log. The Federal Communications Commission requires the following items:

1. An entry of the time each station identification announcement (call letters and location) is made.

2. An entry briefly describing each program broadcast, such as "music," "drama," "speech," etc., together with the name or title thereof, and the sponsor's name, with the time of the beginning and ending of the complete program. If a mechanical record is used, the entry must show the exact nature thereof, such as "record," "transcription," etc., and the time it is announced as a mechanical record. If a speech is made by a political candidate, the name and political affiliation of such speaker must be entered.

3. An entry showing that each sponsored program broadcast has been announced as sponsored, paid for, or furnished by the sponsor.

4. An entry showing, for each program of network origin, the name of the network originating the program.

In addition to this, the station itself may require certain items, such as an entry by the operator that he actually heard each commercial presented. This information is used by the accounting department of the station.

The program log can be prepared by the operator from the daily operating schedule of the station. An example of an operating schedule is shown in Fig. 35. Sometimes the operating schedule has space on it for the operator to write in the proper log entries.

EQUIPMENT MAINTENANCE

Control room equipment maintenance consists mainly of checking the tubes of all amplifiers, regularly cleaning the mixer controls and relay contacts, and making periodic frequency runs with pure tones to check the fidelity of frequency response of the entire system.

Most stations have a regular maintenance schedule that includes keeping a record of the date tube checks are made, of the condition of each tube at the time of checking, and of which tubes are replaced and why.

Mixer controls are cleaned by removing the covers and using carbon tetrachloride on a brush to clean the contacts before they become noisy.

Frequency runs are made by patching the output of a pure tone oscillator into each preamplifier of the channel to be checked and making notes of the VU meter readings for frequencies from about 30 to 8,000 cycles. Frequently these test runs are checked on through to the transmitter to check the line and line equalizers between studio and transmitter. Each station has its own set procedures for maintenance schedules; if you become an operator, you must learn and follow the system used in your own station.

Lesson Questions

Be sure to number your Answer Sheet 42RC.

Place your Student Number on *every* Answer Sheet.

Most students want to know their grade as soon as possible, so they mail their set of answers immediately. Others, knowing they will finish the next Lesson within a few days, send in two sets of answers at a time. Either practice is acceptable to us. However, don't hold your answers too long; you may lose them. Don't hold answers to send in more than two sets at a time or you may run out of Lessons before new ones can reach you.

1. If a pre-amplifier, having a 600-ohm output, is connected to a microphone whose effective output is —40 db, and the mixer loss is 10 db, what must be the gain in order to obtain a +10 db output? (Assume that the input impedance is also 600 ohms.) 60 db

2. If a 20-ohm pick-up is rated at —67 db when connected to an open circuit, what will be its effective output level when it is matched to its own impedance in an operating circuit? −56 db

3. If the effective output level of a microphone is —56 db, and the output of the amplifier to which it is connected is +30 db (1 watt), what is the overall gain of the amplifier? 86 db

4. Why is a dynamic microphone generally preferred for remote broadcasts where the microphone must be handled?

5. Which type of ribbon microphone, *velocity-operated* or *pressure-operated*, has the better response at low frequencies?

6. What is the purpose of a pre-amplifier?

7. Should the microphones in a "dead" studio be placed *closer to, farther from,* or the *same distance from* a sound source than in a "live" studio?

8. In a multiple microphone pick-up, how should the microphones be placed with respect to each other?

9. Why is it generally not necessary to "ride gain" on records and transcriptions?

10. Why is it necessary for the control operator to monitor by ear as well as by watching the VU meter?

HOW STRONG IS THE CHAIN?

Let's suppose you are running a construction job and need a chain to carry a heavy and very valuable load. You order your blacksmith to forge a chain; you want it made of the best steel—each link must be perfect. Such a strong chain will carry the valuable cargo safely. But if just one link parts — DISASTER!

Life is like that. It is like a chain composed of ambition, perseverance, character—and TRAINING. Until your link of Training has been forged there is nothing to tie the Success chain together. And you need a strong link to hold it together.

You are using good, strong, time-tested material now. But don't skimp on material. Don't think you know a subject — KNOW THAT YOU KNOW IT! Don't try to cover a subject or a lesson in a day when common sense tells you it should require a week. Don't ruin your steel by allowing flaws to creep in which may eventually weaken your chain.

Put all your NRI Training into your success chain—make it as big and as powerful as possible. The time will come when you need a strong chain to support you—to help you over difficulties—to carry you through competition—to carry you onward, forward, and upward to the success goal which you have set for yourself.

J. E. SMITH