second edition

BROADCAST OPERATOR'S HANDBOOK

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JOHN F. RIDER PUBLISHER, INC.
480 Canal St. New York 13, N. Y.

FIRST EDITION, FIRST PRINTING; NOVEMBER, 1947
SECOND PRINTING; FEBRUARY, 1949
THIRD PRINTING; SEPTEMBER, 1949
FOURTH PRINTING; JUNE, 1950
FIFTH PRINTING; APRIL, 1951
SECOND EDITION, FIRST PRINTING; SEPTEMBER, 1951

PRINTED IN THE UNITED STATES OF AMERICA

PREFACE TO THE SECOND EDITION

SINCE THE YEAR 1921, which saw the birth of radio-broadcasting as a specialized field, development of almost flawless equipment for transmission and reception has taken place. The surprising aspect is the obvious gap in literature that exists between the field of radio engineering and design, and the practical operation of the products thereof. Probably this is due to the limited number of engineers concerned with broadcast operations; yet there is no subject riper for expansion.

The technical aspects of production and programming, for example, is sparsely covered in existing books or trade journals. This edition has been expanded particularly to cover this field more completely. All other subjects are treated more fully than was the case in the first edition; an outgrowth of innumerable questions from students in basic radio classes, and the many helpful suggestions from interested readers already engaged in broadcast operations.

The subject and content of this book are intended not only for the many newcomers to control rooms and transmitters, but also for the "old timers" familiar with all the problems peculiar to their work. The first four parts cover the operating practice in control rooms, the master control, remote controls, and the transmitter. Parts 5 and 6 are concerned with technical data for operators and engineers.

Throughout the book an attempt is made to bring forth a new approach to modern operating techniques, and to lead to a better understanding between studio and transmitter personnel.

The author wishes to thank the editors of RADIO-ELECTRONICS for permission to use some of the material from his articles appearing in that publication in past years. Chapter 20 contains in part, material appearing in articles by this author in RADIO (now AUDIO ENGINEERING) during January, February, and March of 1945, and is reproduced herein through the courtesy of Radio Magazines, Inc. Some of the material in Chapter 4 appeared previously in the November and December 1948 issues of FM-TV Magazine, and is used through the courtesy of that publication.

Thanks are due to Mr. Joseph Kaufman, Director of Education at the National Radio Institute, for his courtesy in allowing the use of part of the material herein that was written by the author for the new broadcast section of NRI courses.

These acknowledgments would not be complete without extending many thanks to the editors and illustrators of John F. Rider Publisher, Inc. for their patience and invaluable aid in the manuscript preparation. The following concerns have also contributed to the text through supplying photos and/or technical data of their particular products:

Andrew Corp., Electro-Voice, Inc., Johns Manville Corp., Magnecorder, Minnesota Mining and Mfg. Co., Rangertone, H. F. Olson and D. Van Nostrand Co., among many others.

HAROLD E. ENNES

July 18, 1951

Contents

Part 1 OPERATING IN THE CONTROL ROOM AND STUDIO

Part 2 OPERATING THE MASTER CONTROL

Part 3 OPERATING OUTSIDE THE STUDIO

Part 4 OPERATING THE TRANSMITTER

Part 5 WE'RE OFF THE AIR

Part 6 TECHNICALLY SPEAKING

TO
BARBARA JEAN

Part 1

OPERATING IN THE CONTROL ROOM AND STUDIO

Chapter 1

WHAT YOU'RE UP AGAINST

BROADLY SPEAKING, there are three kinds of pickups which concern the control-room operator; namely, studio, remote controls, and incoming network programs where the station is affiliated with a particular network.

Studio programs are all programs that originate at the regular station studios. All the most popular shows, such as Jack Benny, Bing Crosby, and Fred Allen, are studio programs at the main network studios. These same shows, of course, are handled as incoming network programs at the affiliated stations.

A remote control, or "nemo" in radio language, is a pickup originating somewhere other than the stations' regular studios, such as a sporting event or night club. Remote controls will be discussed in Part 3 of this book.

For studio programs, microphones must be set up or "spotted" in the studio in such a manner that all musical instruments and performers that are part of the production will be adequately covered. Sound waves striking the movable element of the microphone causes a vibration in the magnetic field in which the element is suspended, which in turn results in an electric potential on the element varying in accordance with the sound waves. The mechanical construction of various types of microphones used for broadcasting is explained in Part 6 of this handbook. The electric energy thus generated is very weak and must be amplified to an amount sufficient to be carried by wire lines to the transmitting plant. (This is true in all cases except in some of the lowest powered installations where the transmitter and control rooms are installed together. Even when this is true, the signal must be amplified considerably to drive the speech input stages of the transmitter.) Control of the various microphones is provided by grouping individual switches and volume controls for each microphone on a panel known as the *control* console. A volume indicator must be used to indicate the relative magnitude of the program signals and is mounted in a convenient visual area on the control console.

Control Operator

One duty of the control operator is to place the microphones in the studio where the production director wants them for a particular program. Where no production director is employed, the control operator must determine the positions of the microphones, or, perhaps more correctly stated, he must determine the positioning of the performers in the microphone pickup area. The best positions are usually determined only by rehearsing the show before air time, and alternating the respective positions until the proper pickup is achieved.

During the progress of a studio show, the studio operator's position is at the control console. It is his function to operate the various microphone controls so that their respective outputs properly "blend" into the effect desired. When a production director is employed at a station, he will assist the operator by telling him which sound or sections of sound to "bring up" or "lower." Since, in any transmission system, definite limits exist as to maximum volume that can be han-

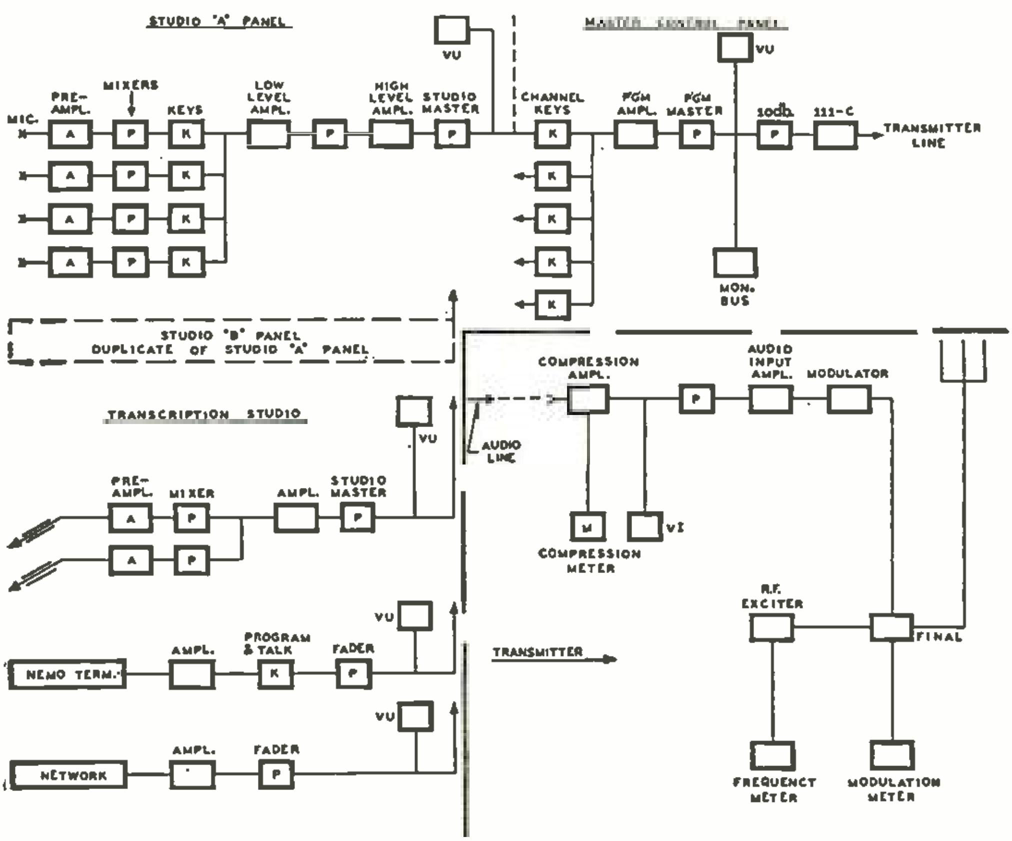


Fig. 1-1(A). Block diagram of a typical broadcast station installation, showing various setups for switching to different studios and remote and network lines, monitoring, etc.

dled and minimum volume that is adequate for transmission, the over-all volume must be monitored and controlled by the operator. This is the purpose of the volume indicator. The operator must also operate the switching system to choose the proper studio or incoming program lines. Technical features of typical control consoles and switching systems are considered in Part 6 of this handbook.

A good studio operator must not only be familiar with the technical equipment, but also be very sensitive to art as well as science in broadcasting service.

Studio and Transmitter Installation

In order to visualize more clearly some of the discussion to follow, it will be helpful to refer to the block diagram of a typical broadcast installation as shown in Fig. 1-1(A). Most of the illustration is self-explanatory, showing in simplified form the setup necessary for mixing

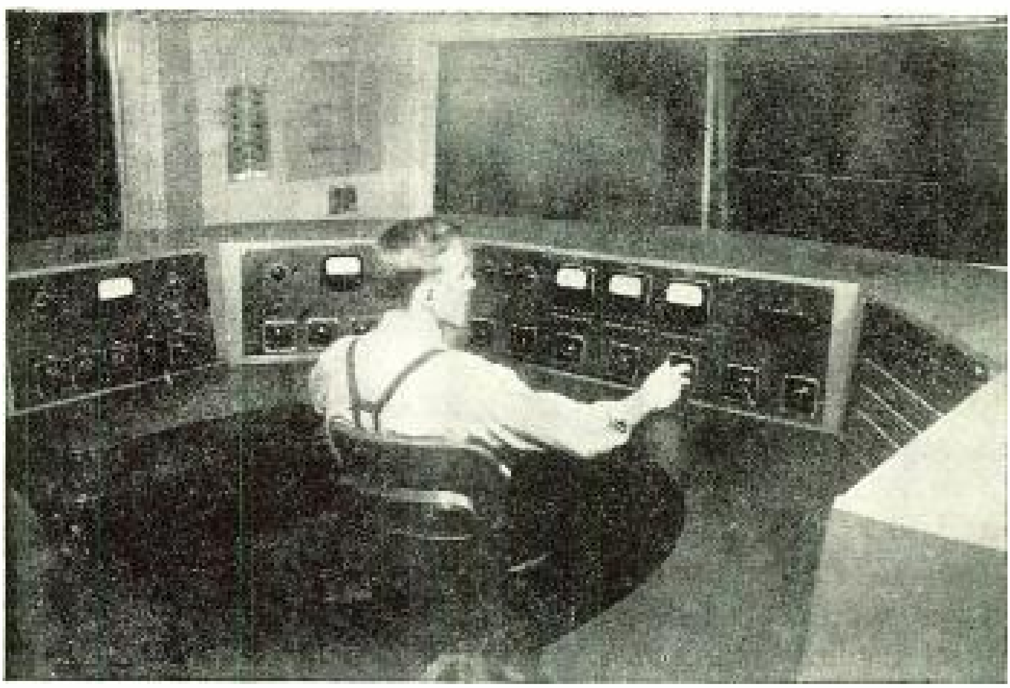


Fig. 1-1(B). The control console of Fig. 1-1(A). The master panel is in the middle with studio panels on either side.

and blending of voice and music from a specified studio, switching of studios, "remote" and network lines, visual and aural monitoring facilities, wire transmission to the transmitter, and associated equipment. A photograph of this centralized control installation is shown in Fig. 1-1(B).

The pad *P* shown in the master control panel circuit before the *111-C* repeater coil is inserted to provide a constant load at all times for the channel amplifier, and is necessary since the equipment sometimes operates at a higher level than is deemed advisable to feed into program lines of the telephone company. The new standard vu meter bridged across the monitoring points are supposedly indicators of 1 milliwatt of power (sine-wave) in 600 ohms. Actually, the meter indicates 0 vu with a sine-wave power at 1000 cycles of between +4 vu and +26 vu, depending on the external resistance used in series with the meter to allow greater bridging characteristics, and to facilitate adjustments to correlate readings of the meters used at slightly different volume levels in the circuit. Rms meters of greater sensitivity have not proved practical to date. It should be kept in mind that this calibration assumes a sine-wave signal, and that under actual program material of energy sufficient for 0 vu deflection, instantaneous peaks will exist of several times 1 milliwatt energy, and average power will be a fraction of 1 milliwatt.[1]

As may be observed from the block diagram, visual indication of the program in progress is provided on the studio panel, the outgoing channel amplifier, the line amplifier at the transmitter, and the final result, monitoring of percentage modulation of the transmitter. The duties of the control operator include not only the "spotting" of microphones for musical and dramatic pickups, switching of studios and lines on scheduled time or cue words, and proper mixing of voice and music on studio setups; but also making certain that his "reference volume" or zero volume level does not exceed that point to which 100% modulation of the transmitter is referred.

"Zero volume" level is simply an arbitrary point, and is not to be thought of as rigid fundamental electrical units of power, current, or voltage. It is necessary that it be understood only in relation to the electrical and dynamic characteristics of the meter used and the technique of reading its response. Perhaps this will be clarified by Fig. 1-2, showing the response of two typical volume indicators on a sudden applied signal. This difference in dynamic characteristics of the new and old type volume indicators (Fig. 1-3) shows the need for a difference of technique in using the interpretation of the meters. The standardization of the new type indicator is a great step forward in broadcasting and most stations are equipped with these meters today.

[1] Chinn, H. A., Gannett, D. K., and Morris, R. M., "A new standard volume indicator and reference level," *Proc. IRE,* vol. 28, pp. 1-17, January, 1940.

It must be remembered, however, that modulation monitors at the transmitter must necessarily be of the semi-peak reading type since this is specified by the FCC (Federal Communications Commission); whereas, the vu meter used at studios is meant to integrate whole syllables or words. This meter is slightly underdamped, which tends to cause the pointer to pause for a moment on the maximum swing, then start downward more slowly than in the case of the previous indicators. Therefore the meter appears to "float" on the peaks without any erratic jumps. The psychological effect is excellent and the meter

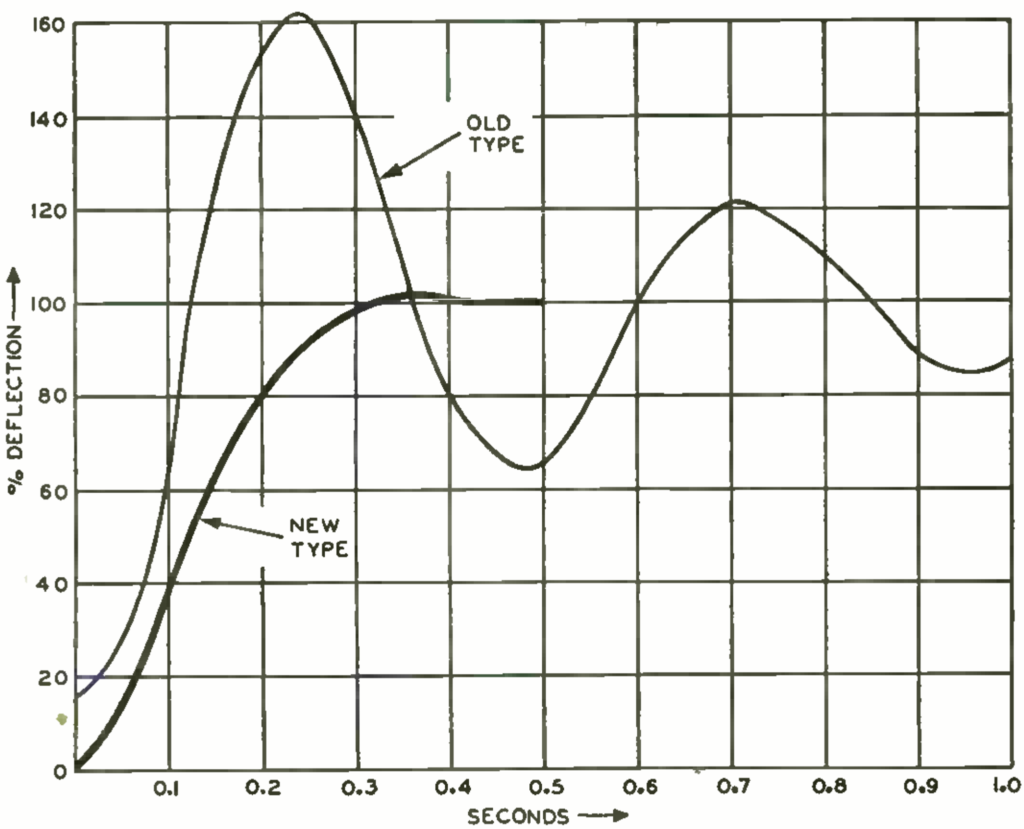


Fig. 1-2. The response of old and new types of volume indicators to a suddenly applied signal indicates a need for a different technique in the interpretation of the readings. *Courtesy Proc. IRE*

appears to show the audio wave as it sounds to the ear from a monitoring speaker. A typical transmitter modulation meter reaches 100 on the scale in approximately 0.09 second when a 1000-cycle voltage of the required amplitude is applied to the equipment; whereas the vu indicator reaches 99 in 0.3 second, as indicated in Fig. 1-2.

Coupled with this difference of dynamic characteristics of the two meters is the conventional habit of monitoring at the transmitter on a single positive or negative peak. By studying Fig. 1-4, which is a graph drawn from a typical oscillograph of a speech wave, it is noted that the energy in positive and negative peaks is far from equal. This is typical of speech waves at the output of a microphone regardless of the type or make of microphone used. Since the vu meter works from

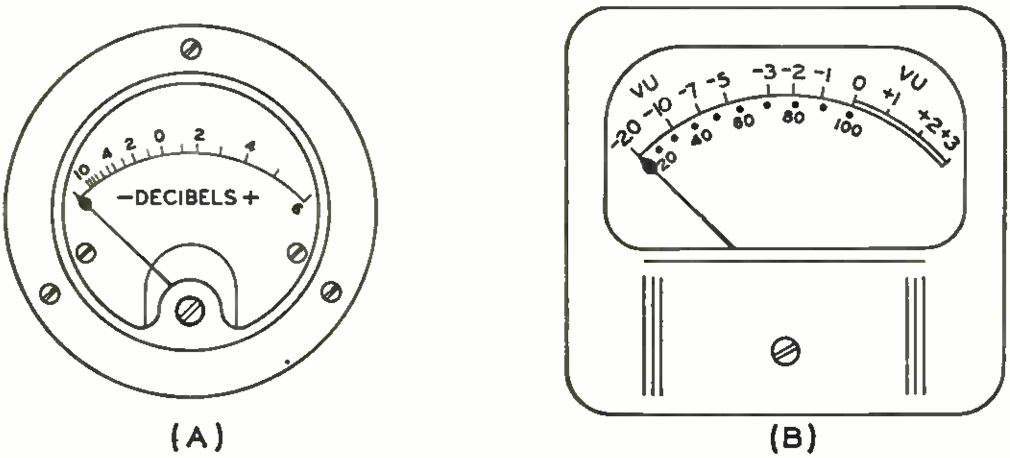


Fig. 1-3. The old type of volume indicator is shown at (A) and the new type at (B).

a balanced full-wave rectifier, its reading is not dependent on the pole of operation, and thus the comparison of the indication at the transmitter modulation meter position with that at the studio cannot be expected to agree even with perfectly matched circuits between.

This one fact is probably the most universal reason for friction between transmitter and studio personnel. It is not possible, for instance,

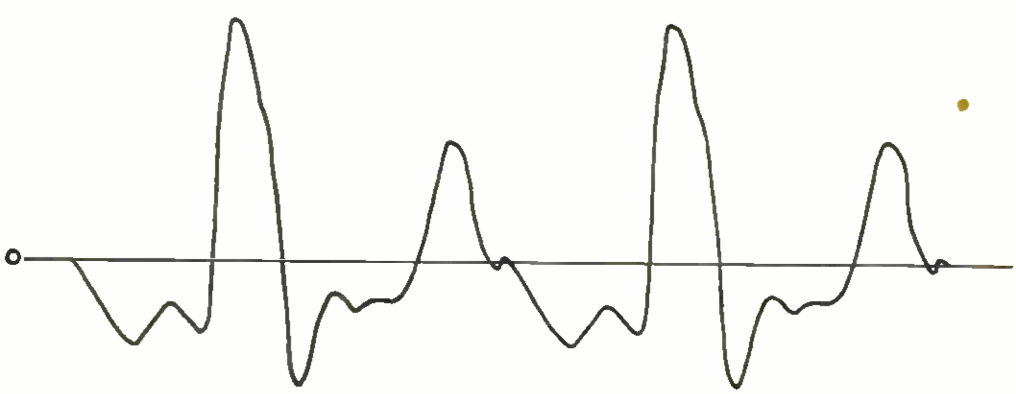


Fig. 1-4. This representation of a typical speech wave shows that the energy in the positive and negative peaks are unequal.

to obtain the same polarity of maximum energy from the two sides of a bidirectional microphone. An interviewer may show an indication at the transmitter of 100% modulation and the interviewee on the opposite side of the microphone (therefore oppositely poled at the microphone transformer) may show only 50% (or less), yet the indicator at the studio (full-wave rectification) will show exactly the same peak level. It is perfectly plausible, then, that the transmitter operator, unfamiliar with speech-wave characteristics through a microphone, should conclude from his monitor reading that the two voices are not balanced at the studio end. This belief is sometimes further encouraged by the extreme difference of "loudness sensation" between two voices of different timbre that are peaked the same amount on a full-wave rectifier indicator. This discrepancy between level indication on a meter and the aural "on air level" is one of the most perplexing problems of broadcasting and will be taken up in more detail presently.

Transmitter Operating Technique

The foregoing discussion brings to mind several questions as to transmitter operating technique. Is there any true indication at the transmitter of comparative levels from the studio? Which pole of the modulation envelope should be monitored continuously and why? These problems, together with detailed suggestions of pre-program level checks will be discussed in Part 4 on transmitter operating practice.

As will be discussed under the transmitter section, it is highly desirable to have all the unidirectional microphones, in use at the studio, poled so that the maximum energy pole of operation will coincide with the positive side of the modulation envelope at the transmitter. The ratio of peak energy difference varies with type and make of microphone, but is apparently most pronounced in pressure-type microphones, such as the RCA Type 88-A. This type microphone, due to its light weight and rugged construction, is often used for remote pickups, and in studios, such as transcription booths, used mainly for announcements. The operator should always take advantage of this characteristic when this type microphone is used, as the results of proper polarization at the studio are very much worth while.

You're up Against Sound

The control operator must be familiar with one of the most complex, and perhaps one of the most controversial scientific fields, the field of sound. Since he must correlate the known facts of sound with operational technique of equipment, it is well to take a little time here with a brief review of fundamentals.

Whenever the air is set in motion at a vibratory rate that happens to fall within the audible range of the human ear, *we hear sound.* This sound, in the strict physical sense, is a setting up of vibrations in the air consisting of alternate condensations and rarefactions of the air constituting *sound waves.* Yet how primitive, how incomplete is this bare definition of sound! The great number of engineers and technicians concerned with the transmission and reproduction of "sound" find themselves faced with problems of unwanted *noise* in sound, of *loudness* in sound, of *tone color* in sound, of the more complex *timbre* of sound. He may read somewhere that it takes about five seconds for a sound wave to travel a mile in air, that it travels four times this fast in water and almost fifteen times as fast through iron, but this picture of sound does not help him in any practical manner. He is taught in his physics courses that loudness depends on the amplitude of the air disturbance, that pitch depends on the vibratory rate of the sound wave, but, in practice, he finds many more variables than these fundamental truths of loudness and pitch. He soon becomes aware of a decided difference between *loudness* of sound and *volume* of sound!

Essentially, the technician or student reading this text is interested in the best possible methods of transmitting or reproducing sound, and the proper ways of care, maintenance, and repair of the equipment involved. He knows that the sounds he hears in nature are very definite and distinct in themselves, that the rustle of dead leaves sounds different than the rumble of thunder. He is able to tell the difference between one man's voice and another, and even to interpret the very spirit that lies behind the inflections of a spoken word. The amplifier with which he is concerned in this problem must be "fed" with electrical impulses that may again be reconverted to all these original characteristics that are so much a part of the original sound in nature. This section is, therefore, dedicated to the man who wants to understand the theory and practice of sound conversion and reconversion, so that he may be as competent with the so-called "gateways" to sound as he is with the amplifier.

The Vivisection of Sound

We usually think of *sound* as something we *hear.* Since this is the most practical approach to a highly complex phenomena this discussion will approach the subject in relation to the effect of vibrations in the air as they affect our hearing sense. This is in spite of the premise that the physicist and the laboratory technician consider the word *sound* to define vibrations in the air or solids that are set up from the vibrations of the original sounding body even though there is no ear to *hear* that sound. The practical man, however; the broadcast technician, the recording operator, or PA operator wants to be able to visualize the nature of his equipment as it affects the ear of the listener. For example, what could be the various causes of amplitude distortion in a microphone? What would affect the frequency response? Knowing the characteristics of a particular microphone when it is in normal operating condition, how can these particular characteristics best be used to achieve the utmost possibilities within its natural limitations? In order to gain the "feeling" and the "know-how" with sound-converting apparatus, we will start with the fundamental nature of sound and the problems associated with their conversion to electrical impulses in such a manner that the ear will be able to hear the reconverted sound as an almost exact duplicate of the original impulses. Sound, of course, may be transmitted in liquids and solids equally as well as in gaseous matter; but this text will be concerned with the nature of sound waves in air, which is the primary concern of the reader.

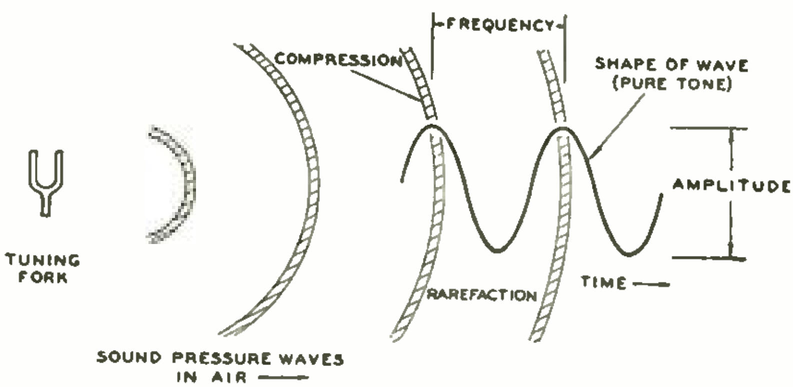


Fig. 1-5. The air compressions and rarefactions produced by a vibrating tuning fork when plotted with relation to time, yield the transverse waveform.

Fig. 1-5 illustrates the fundamental principle of a sound wave. When a tuning fork is struck, it is set into vibration at an amplitude depending on how hard it was hit and at a frequency depending upon the physical dimensions of the fork. As it vibrates, the air surrounding it is disturbed and alternate condensations and rarefactions of the air occur as shown. The "waves" travel in all directions from the fork and decrease in amplitude as the square of the distance from the vibrating fork. If these variations of the air pressure are charted in respect to time, a graphic representation known as the "curve" or "shape" of the sound wave is obtained. This is also shown in Fig. 1-5, where the single "pure tone" of the tuning fork may be observed. The amplitude of the sound wave is represented as the over-all height of the curve, and the frequency, by the distance between successive peaks or troughs of the curve plotted against a definite time axis. For example, if the time occurring between the successive peaks shown in Fig. 1-5 was one-thousandth of a second, the curve would represent a tone of 1,000 cps.

Insofar as pure tones of this kind are concerned, such as those from a tuning fork or audio oscillator, it may be stated that the *intensity* of the sound is dependent upon the height or amplitude of the wave, and the *pitch* of the sound is dependent upon the number of wavelengths per unit of time or frequency of the wave. Please note here that we have defined the amplitude of the sound wave as *intensity*, not loudness! The *loudness* of any sound wave is a more complex phenomena which will be explained shortly.

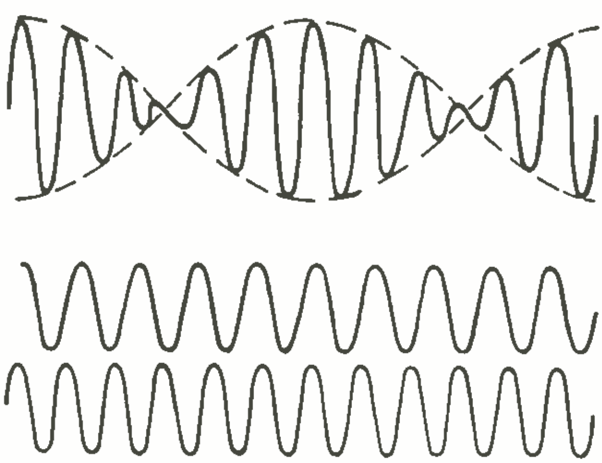


Fig. 1-6. The production of a beat frequency by the addition of two fundamental ("pure") frequencies generated by two tuning forks with different characteristic frequencies.

Now consider what happens when two pure tones are sounded which differ somewhat in frequency. Fig. 1-6 shows two tuning forks, each differing slightly in frequency characteristics, their individual waveshapes, and what occurs to the resultant waveshape when combined in the air. When the phase of these two waves are the same (both rising and falling together) the resultant wave is additive and a large amplitude results. When they differ in phase (one rising, the other falling), the waves tend to cancel out and the resultant amplitude is small. This physical growing and dying out of sound at regular intervals constitute what is known as a "beat tone," and the beats are equal in number to the difference between the two frequencies. For example, if $T_1 = 2{,}093$ cps and $T_2 = 2{,}349$ cps, the beat tone would be 256 cps. In this case, the ear would "hear" not only the two real physical frequencies, but also the "beat" of 256 cps. If a physical analysis of this tone is made, it will *not* reveal any vibration wave component of 256 cps, yet this "sound" will be just as real to the ear as either of the two fundamental tones employed. Such tones have a great influence on tone quality as interpreted by the ear.

This tone quality, or *timbre* of sound is related to the shape of the sound wave. For instance, a flute sounding the tone of middle C (261.63 cps American Standard Pitch) will not sound the same as a clarinet sounding the same frequency. We still recognize one sound as being distinctly that of a flute, and the other sound as being distinctly that of a clarinet. Fig. 1-7 is a graphical integration of these two sound waves, showing why the timbre, or tone quality, differs as interpreted by the ear, even though both tones are of the same frequency.

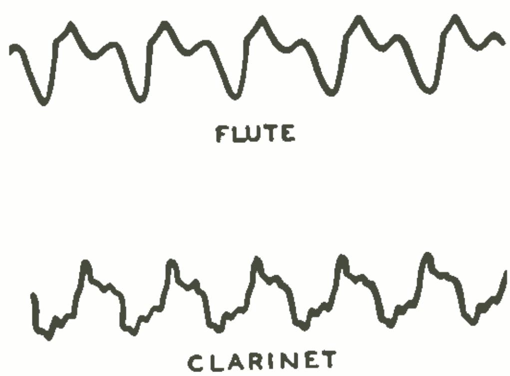


Fig. 1-7. The waveform representing the flute tone is smoother (fluctuates less) than the waveform representing the clarinet tone. This accounts for the difference in tone quality or "timbre," of the two instruments.

Under practical use, a microphone is subjected to a wide variety of tones that make up the character of speech or musical sound. These waves become very complex in character and are termed complex waves. It is the duty of the microphone to convert these complex waves into corresponding electrical vibrations that neither add to nor subtract from the original vibrations of the air particles. When a word is spoken, the sound is given a certain *timbre* or *tone color* or characteristic that is determined by the particular vocal chords, mouth and nose cavities of the speaker. This characteristic of speech is like the fingerprint of a person; no two people have exactly the same voice timbre, and it is the reason why we "recognize" a person's voice even though we are blindfolded. If it is to be used for fidelity, as in broadcasting, any gateway to sound must be able to convey the different shades of expression of the voice or musical sound. In the case of the microphone, it must change vibrations in air into corresponding electrical impulses. In the case of the recorder head, it must convert electrical vibrations into corresponding mass energy sufficient to move the cutting needle. The pickup head must convert the mass energy of the needle in the grooves of the record to corresponding electrical vibrations to excite the amplifier. All of this must be accomplished so that the original shapes of the sound waves which determine timbre or tone quality are maintained as nearly as possible. Under certain conditions, however, this characteristic is undesirable, and we find a need for emphasizing *intelligibility* of voice rather than a wide range response so necessary for musical enjoyment. Details of such apparatus will be discussed in their place.

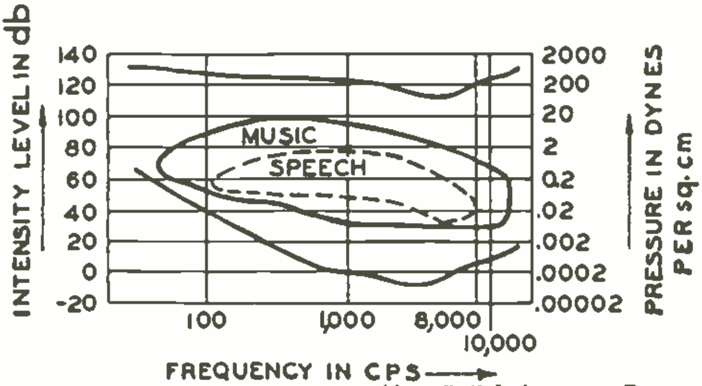
The Phenomena of Hearing

The understanding of any apparatus connected with sound transmission necessitates a thorough knowledge of the human hearing sense. In the previous paragraphs, we have gained a practical working knowledge of the mechanics of sound in air, and what these sound waves must become in order to use them for transmission or reproduction. In the final analysis, however, the ear, or rather the interpretation of the ear, is the basis of all practice and theory in the practical field of sound.

The frequency response of a "good" ear is from 20 to 20,000 cps. The range of the "average" normal ear is generally taken as 20 to 15,000 cps. There is, however, a noted falling off of high-frequency response with advancing age.

When we think of "frequency response," it is well to visualize the effect of differing frequencies on this hearing sense. When it is said that an ear responds to a frequency range of 20 to 15,000 cps, it is *not* implied that the ear responds to a frequency of 40 or 50 cps equally as well as one of, say, 1,000 cps.

Study for a moment the graph of Fig. 1-8, which shows the frequency and volume range of speech and music; the lower curve shows the "threshold of hearing" and the upper curve shows the "threshold of feeling." The lower curve showing the threshold of hearing is a picture of deviation of response of the ear with frequency. That is, where the 1,000-cps tone is taken as zero db and is of such intensity that it is barely audible to the ear, it is shown that for a 100-cps tone



After Bell Laboratory Record

Fig. 1-8. On the curve above, the area marked "MUSIC" and the area marked "SPEECH" enclose those frequencies of music and speech which are intelligible to the naked ear.

to be barely audible it must be approximately 40 db higher than that of the 1,000-cps tone. It is seen here that there is a notable falling off at both low- and high-frequency response when the sound intensity is low. When the intensity is so high that it approaches the threshold of feeling (upper curve), the frequency response of the ear is seen to constitute a comparatively flat line. It is also observed from this illustration that this frequency range of speech is 100 to 8,000 cps, dynamic range (ratio of lowest to highest intensity) of 40 db. The frequency range of music (in this case, orchestra) is about 40 to 14,000 cps with a volume range of 70 db. There is a definite effect of frequency range upon "intelligibility" of speech and quality of speech and music, which will be discussed later.

"Loudness" and the Operator

It should be realized now that "loudness" of a sound will depend not only on intensity but also upon frequency, due to the peculiar properties of the human ear. The radio serviceman is acquainted with the "bass boost" circuit associated with the volume control of a receiver, which boosts the bass response at low volume settings to help offset this characteristic of hearing. In actual practice, it may be noticed by the broadcast operator that two voices peaking the same amount on his volume indicator will sound very different in "loudness." The loudness of a complex wave will depend upon the number of harmonics present in the sound wave, and the phase relationships of these harmonics. Since the timbre of voice waves differ so radically from person to person, the technician should realize that this effect is not an indication of a faulty component, but is a natural phenomena of the hearing sense. It may also be understood from this discussion why a microphone, pickup, or recorder head must introduce negligible phase or frequency distortion since the timbre of sound depends so much upon the preservation of waveshape.

As to the problem of difference in "loudness sensation" for a given meter reading between two or more voices, a brief perusal of the situation will emphasize the magnitude of its importance, and should constitute a challenge to operators and engineers to correlate existing facts with operational procedure.

Fig. 1-9 is a graph of loudness level curves as adopted by the American Standards Association. The derivation of these curves is explained in most standard textbooks on sound and will not be duplicated here. An example will suffice to enable the reader to use this graph correctly. It will be noted, for example, that a tone of 300 cycles, 40 db above the reference level (0 db) corresponds to a point on the curve marked 30. This is then the loudness level. It means that the intensity level of a 1,000-cycle tone (reference frequency) would be only 30 db in order to sound equally loud as the 40-db 300-cycle tone.

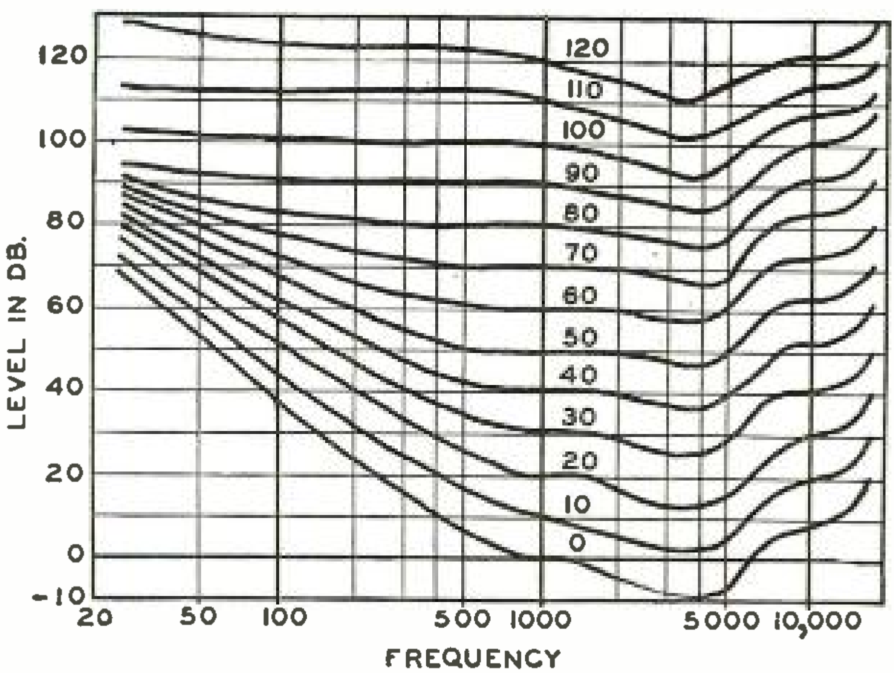


Fig. 1-9. Family of loudness level curves as adopted by the American Standards Association.

Now assuming a fundamental of 392 cycles (G-string of a violin) with an actual intensity of 40 db above reference level, it would be noted from the curve that the loudness level is approximately 36 db.

It was determined in Bell Laboratories that the addition of the overtones or harmonics of the fundamental raised the intensity from 40 to 40.9 db, whereas the loudness level was raised from 36 to 44 db. In other words, the addition of the harmonics raises the actual meter reading only 0.9 db, while the loudness level *increases 8 db*. For the complex tone, the reference level of 1,000 cycles would be 44 db to sound equally loud.

When it is realized that the vocal organs of human beings are all exceedingly different, and are associated with a particular resonating apparatus that gives to the voice its individual timbre, it becomes clear why it often occurs that two voices peaked at a given meter reading will sound far different in loudness. Certain harmonics of the voice are emphasized while others are suppressed in an infinite variety of degrees. A study of Fig. 1-10, which illustrates a graphical integration of two male voices intoning the same vowel, will reveal a decided difference in peak factor (ratio of peak to average content), which in turn depends to a large extent on harmonic content and phase relationship of the harmonics to the fundamental.

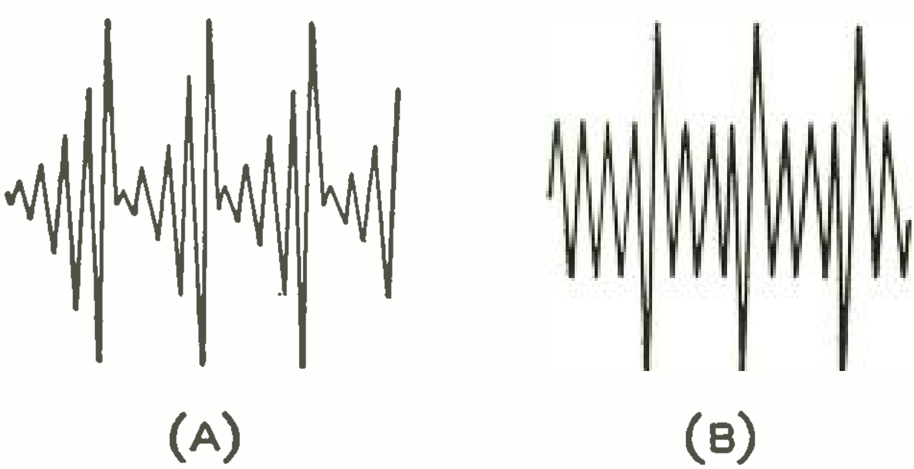


Fig. 1-10. Graphical integration of two male voices intoning the same vowel that discloses a great difference in the peak factor of each.

At the present time only one solution suggests itself. When it becomes necessary to transmit two voices of such difference in timbre as to be decidedly unequal in loudness for a given reference level, the good taste and judgment of the operator at the control panel must govern their respective levels. The author fully realizes the many and varied complications that arise from this condition, since loudness is not only a physical, but also a psychological reaction. The level at which the receiver in the home is operated will determine the extent to which changes in loudness intensity are noticeable, since at low volumes greater change of intensity is required to be noticeable to the ear than is the case at higher volumes. Acoustics of the studio and the room in which the home receiver is operated will influence the ear's appreciation of level intensity changes. However, the control man with good taste, a critical ear, and keen appreciation of values can find the "happy medium" between aesthetics and conventional transmission operations. This is of prime importance when voice and music are to be blended, as will be discussed later.

Narrow Range versus Wide Range

Mention has been made that it is the purpose of a sound-conversion unit to handle this conversion in such a manner that nothing is either added or detracted from the original content, as far as it is possible. There are definite and important exceptions to this rule. A number of transmission systems require a characteristic of *speech* that is known as "intelligibility" rather than fidelity. Some "special events" broadcasting as occur from airplanes and points of high noise levels require such systems. Where quality of speech is not the primary concern, transmission and reproduction may be greatly improved by a narrow band use. This is so because noise and interference levels increase with bandwidth. It has been found that a restricted bandwidth of 400 to 3,000 cps will give maximum intelligibility of speech (with greatly reduced quality) and many units are designed for such use, as will be discussed in detail in the part of this text devoted to voice range microphones.

It is, of course, apparent that quality and artistic values of either speech or music are seriously impaired by this restricted frequency range. Most speeches and all plays need the full voice range of 100 to 8,000 cps. Music, for full appreciation of its beauty and shades of expression, needs the entire frequency range of 20 to 14,000 cps. Dynamic range also is an extremely important necessity for music.

Details of Control Room Metering Circuits

A few of the fundamental characteristics of broadcast metering circuits have been discussed in relation to the proper understanding of their functions by both control room and transmitter personnel. The operator employed in broadcasting is confronted with the handling and measuring of the quantity known as the "volume" of sound. His conception of volume must necessarily be influenced by other than precise mathematical relationships of electrical units such as power, voltage, or current. At the same time, his means of measuring the complex and nonperiodic speech and program waves must be based primarily on a-c theory in terms of the related values of sine-wave currents. The correlation of data on program metering circuits to serve definite performance characteristics for various parts of a transmission system is a most important subject and yet perhaps the least understood among the operators and technicians concerned with their use.

Since the earliest days of electrical program transmission, about 1921, when it became apparent that distortion due to overloading of an amplifier was far more noticeable in a loudspeaker than in the ordinary telephone receiver, various schemes for measuring the magnitude of program waves have been developed. The first device was a d-c milliammeter connected in the output of a triode detector, with an input potentiometer to adjust the sensitivity in 2-db steps. Thus, by adjustment of the sensitivity control so that peaks of program waves caused an indication of approximately mid-scale on the meter at intervals, and by operating the telephone repeaters at about 10 db lower on peaks than the point of overload, a visually monitored program circuit became a reality.

From this early start, there developed a long series of devices, some with tube rectifiers, others with dry rectifiers (peak and rms indicating), full- and half-wave rectification, all degrees of damped movements, and calibrated with reference levels of 10^{-9}, 1, 6, 10, 12½, or 50 mw, in 500 or 600 ohms impedance. It was not until 1938 that the Bell Telephone Laboratories, the Columbia Broadcasting System, and the National Broadcasting Company pooled their knowledge and problems in a joint effort to develop a standardized volume indicator with the reference level implied in the definition of volume units.

The outcome of this concentrated study was the new standard vu meter such as those on the control panel of Fig. 1-1(B). The schematic diagram of Fig. 1-11 shows the circuit used to bridge the meter across program lines or individual studio output lines. It is seen that the total impedance presented to the line is about 7,500 ohms; 3,900 ohms in the meter and about 3,600 ohms supplied externally to the meter. The dynamic characteristic is also standardized as mentioned earlier, being such that, if a 1,000-cps voltage of such amplitude as 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 seconds and overswing the 100 mark by not more than 1.5%. This meter is a great improvement over previous volume indicators where large amounts of overswing often occurred.

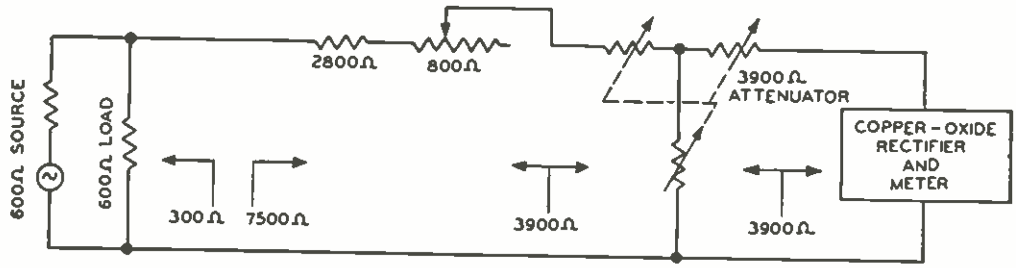


Fig. 1-11. Circuit that is used to bridge the new vu meter across program lines or individual studio output lines.

The graph of Fig. 1-4 is a representation of a speech wave taking place in the time interval of 1/100 second. It is obvious that due to mechanical inertia of meter movements, true "peak reading" indicators are impossible in the strict sense of the term. Since the copperoxide instrument, indicating rms values, is sufficiently sensitive without the use of vacuum tubes, and since in the final analysis, the indication on the meter should follow as nearly as possible the psychological effect of hearing, this type indicator has been standardized for control room and line transmission use. Although peak-reading indicators are much faster than rms instruments, in actual practice their accuracy is limited to about 100 cps. At the higher frequencies their function is to integrate the speech occurring over a period of time.

As will be discussed in detail in the transmitter section of this handbook, the FCC specifies some form of semipeak indicating meter to measure modulation percentage at the transmitter. This is necessary since the peak factor (ratio of peak to rms values) of program waves may be 10 db or more, and when these peaks occur in rapid succession, danger of breakdown in circuit components exists as well as the occurrence of adjacent channel interference. Means must also be provided to check positive or negative peaks of the modulated carrier. Thus the modulation monitor is essentially a half-wave rectifier indicator. These differences between studio and transmitter meter characteristics must always be borne in mind by the operators.

Volume Indicator Interpretations

For the purpose of discussing the problems relating to use and interpretation of volume indicator readings, the well-known fact that any wave, no matter how complex, may be reproduced exactly by a number of sources of pure tones, will be employed.

Fig. 1-12(A) shows two simple harmonic motions of the same frequency, differing slightly in phase; that is, tone *b* is lagging behind tone *a* by a certain number of degrees. The vector addition at the right shows how the total amplitude is influenced by the reinforcement of the two tones, causing an addition to the total magnitude over either one by itself. Fig. 1-12(B) illustrates what happens when the same tones of similar frequency are differing in phase by exactly 180°. The vector at the right shows complete cancellation of the total energy, since the tones are now opposing each other resulting in zero amplitude. Keeping this clearly in mind, it is seen that the total amplitude will be larger for smaller angles of phase difference, and if the two tones are exactly in phase, the parallelogram at the right collapses and becomes a straight line, that is, the sum of the individual amplitudes. As the angle of phase displacement becomes larger, the total amplitude becomes less, until at 180° it becomes zero.

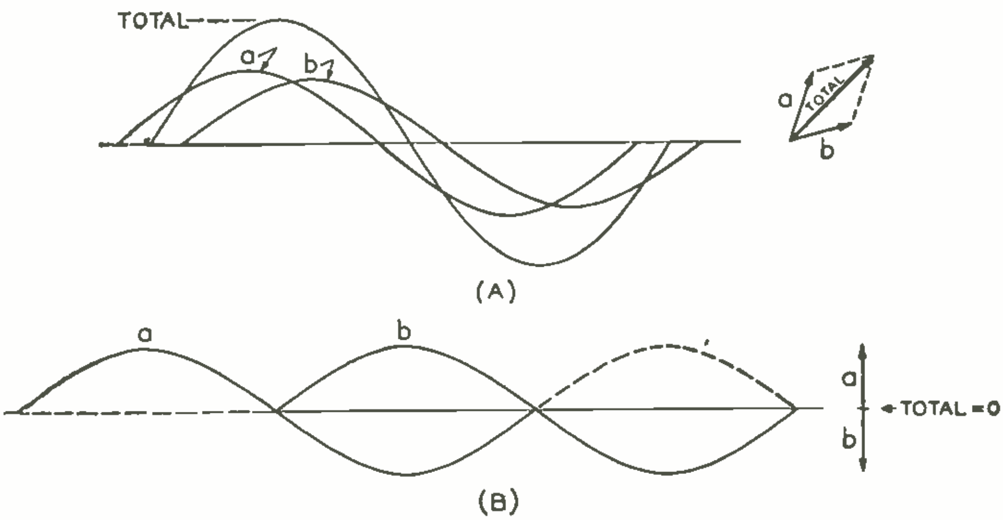


Fig. 1-12. When two harmonic motions of the same frequency, waves *a* and *b*, are out of phase by less than 180°, their total is greater than either, as shown by the vector at right of (A). When *a* and *b* are 180° out of phase, they cancel as shown in (B).

In actual practice, it is realized that under program conditions a large number of different frequencies with varying phase displacements are being covered, and the loudness sensation produced in the ear for a given meter reading is dependent on the number of harmonics present and the phase relationships of these harmonics. It then becomes obvious that acoustical treatment of studios and type of program content will influence the correct interpretation of a volume indicator reading. It is, of course, apparent that the volume indicator performs its duty in respect to showing the magnitude of the waveform, whether it be "distortion peaks," noise, or musical sound, that must be kept within the dynamic range of the transmission system. But when correlating the reading of the volume indicator with the effect produced on the hearing sense of the listener, these problems must be met and analyzed. The loudness sensation for a given meter swing indication has already been discussed in regard to voices of different persons. The same characteristic is noted between individual musical instruments where the number of harmonics and their phase relationships may vary in wide degree.

Dynamic Range Indication

The volume indicator is used as a means of visually monitoring the magnitude of the program waves for two primary reasons: (a) to compress the original wide dynamic range to an amount consistent with good engineering practice of the broadcast transmission system, and (b) for locating the upper part of the dynamic range below the overload point of associated equipment. For the latter purpose, a scale of 10 db would be adequate; for the former purpose, a much wider decibel range is desirable. Since the instrument is used for both applications, it was decided that a compromise on a scale length of 20 db would be desirable. It appears that as the art and appreciation of higher fidelity service advances, not only the frequency range but also the dynamic range of transmission will be extended, particularly for certain types of program material. This feature becomes a very important one for f-m service. With present-day meters, it is left to the experience and judgment of the operator as to how far the volume should be allowed to drop below the visual indication of the vu meter.

It would appear then that, in the interest of good operating practice, some form of auxiliary monitoring of passages below the present-day meter indication point on types of programs requiring wide dynamic range, might be worth while. A suitable oscilloscope used in conjunction with the meter might be one solution. The practicability of this method would be doubtful due to the fact that the operator's attention must be divided between the monitoring device and the studio action. The same drawback would exist for an indicating meter of such wide scale that the entire meter action does not fall readily into the operator's line of vision. A meter of, say, 270° scale would have the same disadvantage of having either the low-passage or maximum-passage indication fall at an awkward position.

Perhaps the most practical solution would be a device consisting ssentially of two indicating pointers, one immediately below the ther. The lower pointer would indicate the first 50% of channel tilization, with automatic overload protection built into the movement (by limiting tube or network) to prevent overload of this movement when the volume is such that the upper movement is indicating he final 50% of channel utilization. The design should be such that ither indicator is in a practical plane of vision for the operator.

Chapter 2

ARE MECHANICAL OPERATIONS APPARENT?

IN THE REALM of the physical, mental, and psychological faculties of the control-room operator lies the success or failure of the broadcaster's daily schedule. A script-writer's masterpiece or a composer's dream can amount to no more than the original worth of his work, plus the ability of the control man to interpret that work on the technical equipment at his command. Yet, perhaps paradoxically, the best qualified operators are the least conspicuous to the listener at home.

An ideal job of switching and blending of microphones for the various performers of a given show is such that the listener-in is entirely unconscious of mechanical operations necessary to their performance. The operator who "cuts" his program at its conclusion, instead of fading out (even though because of time limitations it must be a "quick fade"), not only makes mechanical operations apparent to the listener, but marks himself as a man not entirely a master of his equipment. Exceptions to this rule exist, such as "stunts" of a technical nature that are sometimes aired to impart a technical flavor to the layman. In such programs, technical operations should be accentuated, of course, rather than subdued, but it is well to remember that, as a general rule, the test of the operating technique should be, *"Are mechanical operations apparent?"*

This primary rule should govern the entire operating technique of the control man. Music, speech, or background accompaniment that is too high or too low in level should be gradually adjusted to normal in a manner cognizant of musical and dramatic values. It might appear at first that this prime requisite for good operating practice would conflict seriously with good engineering practice. When levels are too high, overloading of associated equipment at the transmitter occurs. When compression amplifiers are used, as is commonly the case, the distortion arises from excessive compression rather than from overmodulation of the transmitter. It has been proved from extensive tests, however, that distortion caused by momentary overloads simply is not noticeable even to highly trained ears. This appears to be due to physiological and psychological factors that determine the ears' appreciation of aural distortion, resulting in a lack of response to overload distortion occurring at rare intervals and of short duration.

The level of the speaking voice can be least obviously adjusted by correcting the fader setting between words or sentences where slight pauses occur, rather than increasing or decreasing the volume during

PROGRAM TITLE : Musical Clock   FOR : Books   MEDIUM: ET and Recordings

DATE OF BROADCAST : Thursday Aug.19   FROM : 7:15 am   TO: 8:15 am

RECORD OR TRANSCRIPTION

TITLE OF COMPOSITION	BRAND	SERIAL NUMBER	PERFORMING SOC.OR LICENSING AGT.
Got the Moon in My Pocket	W	5163	ASCAP
How About You	Vi	27749	ASCAP
Ferry Boat Serenade	W	3921	ASCAP
Back Home Again in Indiana	De	3786	ASCAP
Blueberry Hill	W	3829	ASCAP
America, I Love You	Co	35865	ASCAP
Chinese Lullaby	W	5690	ASCAP

LEGEND: W - World
Vi - Victor
De - Decca
Co - Columbia
Th - Thesaurus

Fig. 2-1. Typical music sheet for a program of recordings supplied to the announcer operating the turntable and the control-room operator. Brand names enable operator to anticipate volume level.

actual excitation of the microphone by the sound waves. A comparison of these two methods by the operator on his audition channel will reveal the striking difference in the obviousness of level control.

Anticipation can play a major part in smooth level control when circumstances permit. The operator soon becomes familiar with the approximate fader setting of each announcer as he takes over to relieve the preceding announcer. It is obvious here, of course, that ample opportunity is given to adjust the mixer gain before actual air time. This is also possible in some instances with transcribed and recorded shows, when the operator is aware of the brand of recording to be played next.

Fig. 2-1 is a reproduction of a musical sheet for a given program as used at Station WIRE in Indianapolis. The announcer, who operates the turntables, and the operator in the control room each has a copy on hand for reference. It will be noted that the brand of each number, such as World, Victor, Decca, is clearly indicated. This enables the operator to anticipate the level to a certain extent, since, for example, World transcriptions are several vu lower in level than Victor recordings, requiring a higher fader setting. This will also be influenced to a great extent by the type of filter used on the turntable for various recordings, since a different filter is used in many instances for different brands or conditions of recordings and transcriptions. This is discussed more fully in the section on turntable operation in Chapter 4. The gain settings for a given brand will usually be fairly consistent. Thus when the operator has become familiar with the necessary fader adjustment for each brand of transcription or recording, he will be able to use the art of anticipation to good advantage. When the level to be anticipated is uncertain, it is well to remember that from an aesthetic point of view as well as a technical point of view, it is far better to be able to "fade in" the speech or music rather than to experience the shock of excessive volume which must be quickly lowered to normal values.

The foregoing discussion is likely to lead to an erroneous point of view to a newcomer in a control room. It might be well to point out at this time that one of the greatest errors of new men in this field is to "ride gain" to the point of exasperation to a critical listener. The operator should endeavor at all times to give musical and dramatic values a free rein insofar as is practically possible. Remember that from the listener's point of view, the business and purpose of broadcasting is to provide entertainment through the medium of bringing music and dramatics into the home. The technical setup necessary for this purpose has been engineered to a point of perfection; it is only necessary that this equipment be operated in a manner that will promote these musical and dramatic values in their original intent.

The fundamental rule of good operating technique is probably the most abused by innocent operators during the transmission of symphony broadcasts. Suppose that an orchestra of some 40 to 80 members has just finished a number which for the past few minutes has been very pianissimo, say -15 to -20 vu. It is safe to say that the average listener to a symphony program will have his receiver volume adjusted so that comparatively high power exists in the speaker when the studio level hits 0 vu. It is obvious then, what will occur if the announcer suddenly pops in at 0-vu level. The listener may not be actually raised from his chair by this sudden human roar, but the experience, to say the least, is a shock to all five senses. Sudden crescendos in music are expected, welcomed, and appreciated, but a single announcer, exploiting the glorious qualities of Joe Glotz's Super Zoot Suits at an apparently greater volume than all 80 men with everything in their possession from a piccolo to kettle drums, simply is not only unwelcome, but extremely obnoxious.

It is a safe rule to remember that after such musical numbers as this, the announcer should be held down to about −6 maximum. The difference to be maintained between levels of voice and music will depend not only upon the type of program aired, but also upon the acoustical treatment of the studio and will be mentioned in Chapter 3.

The inadequacy of present-day broadcasting to the field of symphony music transmission is quite apparent to most engineers. The discrepancy between the usual 70-db dynamic range of a full orchestra and the actual 30 to 35 db allowed by broadcast equipment is all too obvious to the control man handling such pickups. It has been the practice of some operators who do not appreciate the symphonic form, to bring all low passages up to around −4, then "crank down" on the gain as the orchestra increases its power according to the continuity of the musical score. The fault in this technique should be apparent. If the very lowest passages are brought up to just "jiggle" the meter, and care is taken to use good taste in suppression of the crescendos, a very satisfactory dynamic range may be experienced, since even a range of 25 db will vary the output at the receiving point from 25 mu to nearly 18 watts *on peaks.*

Needless to say the technician on a symphony program, or any musical program, should possess a good ear for music. Rules and regulations will never help a man with a pair of "tin ears" to handle a musical show properly. There are, of course, many competent technicians who do not like or appreciate music, and these men should be assigned to the performance of technical maintenance or transmitter duty. It is, nevertheless, important that the transmitter technician understand that a great amount of modulation during classical music will be below 20% even with compression line amplifiers.

Recordings and transcriptions of symphony music have already been compressed into broadcast dynamic range, since the recording engineer has essentially the same problem to contend with in relation to this difficulty. Usually all that is necessary for the control technician to do is to set the level on the peaks of the music to correspond with 0 vu or 100 on the scale, and "let it ride."

Symphony pickups will be more thoroughly discussed in Chapter 4.

Don't Be Hypnotized

The foregoing discussion and the next chapter on keeping sound "out of the mud" should serve to warn both newcomer and oldtimers against one of the most common occurrences in control rooms—the hynotized operator.

A volume indicator is apt to exert a strong hypnotic effect on the person responsible for "riding gain" on a program. So long as the meter "peaks" at 0 or 100% on the scale, the suggestion to the subconscious mind that "all is well" is almost overpowering. in spite of the aural reaction that ordinarily would warn the operator. Competition in the field of modern operating technique demands more and more co-ordination of aural effects and volume-indicator interpretations.

The rule here is to consciously train the *aural response* to command attention over and above the vu meter indication. If a certain voice "sounds" low in volume in relation to music, peak music just enough lower than the voice to compensate for this effect. If two or more voices on the same program "sound" different in loudness, balance them aurally, *then* note the respective vu meter readings to obtain this balance.

It is quite often possible to alter the "loudness" sensation of a particular voice by changing the distance from the mike or angular relationship to the axis of the mike.

Chapter 3

KEEPING SOUND "OUT OF THE MUD"

THE PROBLEM of correlating volume levels with comparative loudness of speech and music has appeared as an item of major importance and should no longer be ignored by broadcast station personnel. Table 1 was compiled as a result of "group tests" of comparative loudness of different types of music with that of speech.[1] The "peak factor" (ratio of peak to rms values) of speech waveform is very great in comparison to that of music waveform, as emphasized by Fig. 3-1. It is apparent, therefore, that 2 to 3 db more power may exist in speech waves in a circuit monitored by an rms meter than is indicated by the meter itself. This will explain the results shown in Table 2, which as well as Table 1, was taken from the aforementioned article. It is apparent then that when speech and music levels are

TABLE 1

Type of Program	Volume Indicator (RMS) Reading for Same Loudness as Speech
Male speech	0
Female speech	0.1
Dance orchestra	2.8
Symphony orchestra	2.7
Male singing	2.0

Showing importance of peaking music 2 to 3 db higher than male speech for equal loudness sensation. (See text.)

adjusted in correct ratio to avoid overloading, the loudness will be approximately the same.

Table 1 contains a discrepancy with the author's personal experience, and is mentioned with the hope of further research and clarification. It will be noticed that results of the tests on this particular group of listeners dictated the need for a 2.8-db higher level for a

[1] Chinn, H. A., Gannett, D. K., and Morris, R. M., "A new standard volume indicator and reference level," *Proc. IRE,* vol. 28, pp. 1-17, January, 1940.

dance orchestra, and a 2.7-db higher level for a symphony orchestra over that of male speech. If the author was to compile a similar table of equal loudness from several years experience of watching volume indicators (VI's) on various types of programs, he would choose approximately 3 db higher level for a dance orchestra, and 4 to 6 db higher for a symphony orchestra over that of male speech. The author

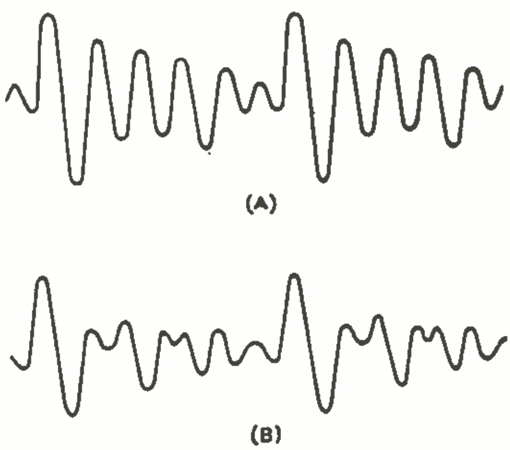


Fig. 3-1. The "peak factor" (ratio of peak to rms values) of music waveform (A) is not as great as the peak factor of voice waveform (B).

feels that this is not caused by a different physical response of the ear itself, but rather to a possible difference of acoustical factors involved, plus the fact that certain psychological factors were not considered in the original tests. By this is meant the important difference in

Table 2

Type of Program	No. of Tests	Total No. of Observations	RMS Volume Indicator
Male speech	8	81	22.1
Female speech	8	82	22.8
Piano	5	40	24.1
Brass band	4	25	24.1
Dance orchestra	5	42	24.7
Violin	1	15	25.8
Average speech	16	163	22.4
Average music	15	122	24.5

The final column shows average overload points of different types of programs, measured at the output of a W.E. 94B amplifier. The important fact of this table is the revelation that the point of overload for average speech is about 2 db lower than the point of overload for average music (rms volume indicator).

listening technique between the symphony audience and the dance-music listener.

As was mentioned before, because of the nature of the classical type of music, the symphony fan at home will operate his receiver on the average a great deal higher in level than he would for ordinary programs. Five minutes of symphony music will have perhaps 3 to 4 minutes of low to very low levels; the average intensity level over a period of time is far lower than the average intensity level of a dance orchestra in the same time interval. It should then be obvious that a greater difference should exist in the ratio of music to speech levels for symphony programs than for those of dance music. Perhaps if tests were carried out with this difference in receiver volume considered, as well as the type of music on the program, the results would be more nearly in agreement with the foregoing argument.

The acoustical treatment of the studio in which the program originates will affect to a great degree the loudness of voice and music, and in a different ratio. A studio that is overtreated with absorbent material deadens the sound because of high-frequency absorption, and is an outstanding enemy to musical programs. Music from "dead" studios is "down in the mud," lacking in brilliance, and generally dull to hear. The effect on speech, however, is not so pronounced as that on music. Speech originates within a few feet of the microphone and requires much less reverberation to assure naturalness, whereas the space

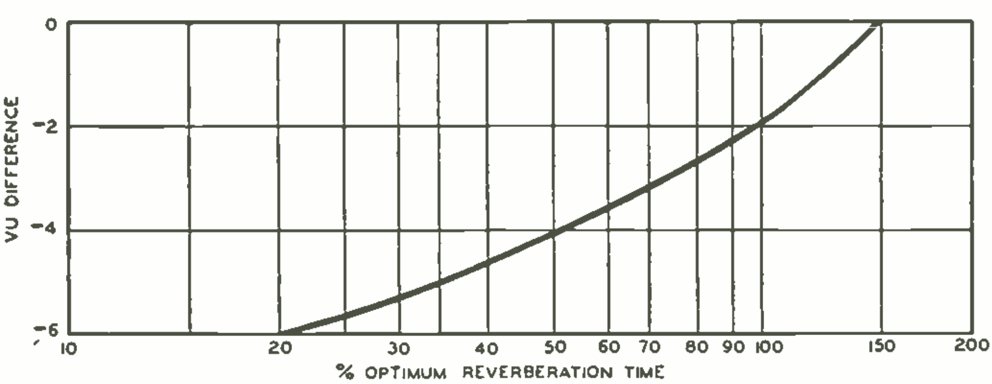


Fig. 3-2. Graph showing the influence of acoustical conditions on ratio of peaking voice and music assuming a necessary 2-db difference for optimum acoustical conditions.

between the source of the music and the microphone is greater, and many things happen to the musical waveforms that must eventually be translated into perceptions of loudness.

Fig. 3-2 is a graph drawn on the assumption of a necessary 2-db difference of voice and music level readings on an rms meter under normal acoustical conditions. The optimum reverberation time will vary according to the size of the studio, as shown by the curve of Fig. 3-3. The curve of Fig. 3-2 is drawn on a probability basis, correlating known facts concerning reverberation time with loudness sensation of voice and music. This graph shows the necessity of a lower peaking of voice in relation to music for less reverberation time than normal,

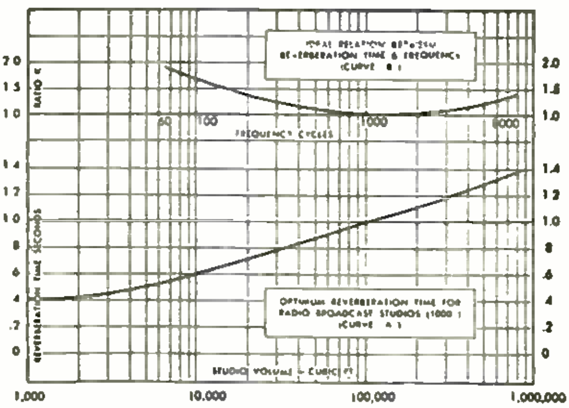


Fig. 3-3. The lower curve (curve A) gives the optimum reverberation time at 1,000 cycles for studios ranging from 1,000 to 1,000,000 cubic feet. The upper curve (curve B) shows the relationship between reverberation time and frequency.

Courtesy Johns-Manville Corp.

and at the same time shows that for 1.5 times the optimum reverberation time, where a great amount of reinforcement of the original musical waves takes place, the voice and music should be peaked the same.

It should be pointed out here that so called "optimum reverberation time" really is an expression of what constitutes pleasing sound, and this conception is still changing with experience. It may well be that near future standards of optimum reverberation time will see a condition which will decidedly alter the above discussion of ratio in peaking voice and music. The point that is important to keep in mind is that a great majority of present-day studios throughout the country are below even up-to-date standards of correct reverberation characteristics; hence the need for the discussion.

The newer "live end, dead-end" studios, with musical instruments placed in the live end and microphones spotted in the dead end, present one solution for properly controlled reverberation. In these studios, voice and music peaked at the same level will appear the same in loudness sensation. In fact, the advancing state of studio development points to all indications that the present day is experiencing a transitional era in which, from some of the most modern studios using reflecting panels for musical pickups, the brilliance of the music is so great that, when peaked an amount on the meter equal to that of

Courtesy National Broadcasting Co.

Figs. 3-4(A) above. 3-4(B). The live end of a "live-end, dead-end" studio designed to provide properly controlled reverberation. Slanted wooden sound-dispersing panels are suspended against the side walls, forming a series of resonance diaphragms, shown in (A). The walls of this studio are slanted (B) to eliminate standing waves that cause flutter.

TABLE 3

Instrument	Studios of Optimum Reverberation Time, Distance in Feet	Studios of 25% Optimum Reverberation Time, Distance in Feet
Bass viol	6	4
Bass saxophone	6	4–5
Trombone	7	5
Trumpet	12	7
Trumpet (muted)	8	5–6
French horn	8	5
Clarinet	8	5
Flute	6	3–4
Violin	5	3
Piano	15	10
Electric organ	15–20	8–10
Pipe organ	20–25	10–15

voice, the voices sound much lower in loudness than the music. This brings to mind again the importance of using judgment in aural perspective when "riding gain" on productions with the intent of achieving a properly balanced effect in the listener's home.

The use of wood in broadcasting in accordance with exact acoustical specifications for controlled reverberation was apparently introduced by CBS in New York about 1935. The entire "live end" of the studio was constructed as a series of resonance diaphragms of seasoned wood, held in suspension with air chambers behind them, as shown in Fig. 3-4(A). The wood panels that cover about one-third of the side walls are placed on slanted surfaces so that the side walls form shallow "V's" running from ceiling to floor. These are so placed that the wall surfaces are not parallel to one another, as shown in Fig. 3-4(B). This eliminates standing waves which would normally produce "flutter." However, because of the highly reflective surfaces of the wood, a certain amount of reverberation is achieved, a quality which adds life and brilliance to the speech and music originating in this type studio.

This is essentially the same principal used by WBBM (CBS) in Chicago and other key points, as well as by the other major networks in their key stations. A number of smaller independent stations have since utilized this type of construction in their studios, and it is hoped that others who contemplate new studios or remodeling of old ones will recognize the tremendous importance of a degree of liveness in broadcast studios.

Chapter 4

YOU'RE OFTEN A PRODUCER TOO

THE OPERATOR of the control room is called upon many times to set up complex musical and dramatic shows. This is especially true in smaller stations that have no production man, and is sometimes true of important key network stations where the control man must achieve the desired results of the production man assigned to a particular show. The responsibility of setups of studio shows is not a simple one. Many years of research and much thought have gone into production, and a knowledge of at least the fundamentals of the art, as they affect the technical duties, will help the control technician over many difficult situations that will arise in the course of his work.

Before taking up actual technical production technique, it is imperative that the operator thoroughly understand the microphone, its possibilities and limitations.

Why Microphones Have "Patterns" of Response

In every different type of microphone there exists a different "pattern" of response. A pattern of response simply means a plot on paper that shows the amplitude response for varying positions of the sound sources about the microphone. Study for a moment the diagram of Fig. 4-1. In 4-1(A) is illustrated what is known as a *nondirectional* pattern; that is, as the sound source is moved around the microphone and kept always at the same distance and intensity, the amplitude of the electrical impulses from the microphone will remain of equal value. Now look at Fig. 4-1(B). Here it is noted that the amplitude of response from the microphone decreases as the angle of the sound source approaches 90° from either "face" of the instrument. This microphone is sensitive on front and rear, and theoretically "dead" at both sides, resulting in a pattern of response termed a *bidirectional* pattern. Still another fundamental type of response pattern is illustrated in Fig. 4-1(C). This is termed a *unidirectional* response, and is sensitive only to sounds originating in the front of the microphone.

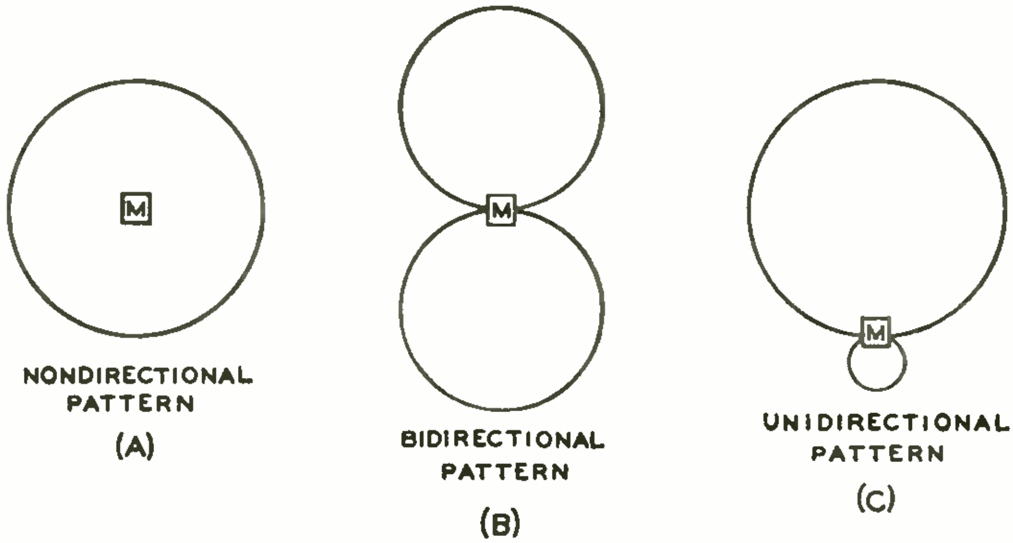


Fig. 4-1. Shown here are the three fundamental types of microphone response patterns.

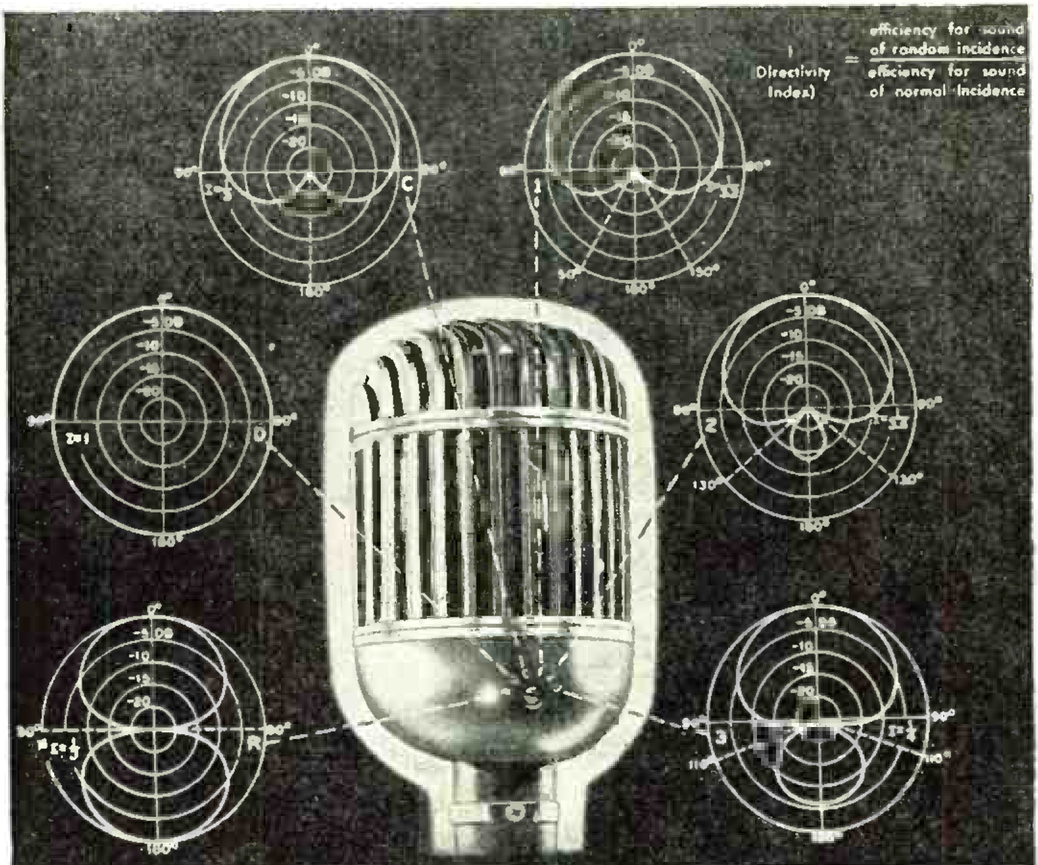
Fig. 4-1(D) illustrates a variable pattern microphone. (Described fully in Part 6.)

The reader should bear in mind one very important note at this point. *These response patterns are calculated in a room so "dead" acoustically that the condition is equivalent to a microphone in open space where there is absolutely no reflection of sound waves from walls, floor, or ceiling.* The effect of room acoustics on the action of the microphone will be thoroughly discussed later in this chapter. It will also be shown that response patterns vary somewhat for different frequencies.

The diagrams of Fig. 4-1 illustrate the three fundamental types of response patterns. These patterns are the result of the design and construction of a microphone and should be understood by the technician in relation to that design and construction.

In the study of any mechanical or electrical component there is always mentioned an "ideal" which the designer must use as a criterion of performance. An ideal microphone, therefore, for most purposes should have an output voltage independent of frequency, and a uniform response for any one direction of a sound wave. In this ideal case, the instrument would deliver power to the audio amplifier whose frequency, phase, and amplitude components would be exactly the same as the original acoustical energy.

There are two basic types of microphone; the pressure type, and the pressure-gradient. Pressure types are quite varied, including the carbon, condenser, crystal, moving-coil (dynamic), etc. There is only one type of pressure-gradient instrument, which is the "velocity" incorporating a metallic ribbon suspended in a magnetic field. There



Courtesy Western Electric Co.

Fig. 4-1(D). Variable-pattern microphone showing the patterns for each of the various switch positions.

are, however, a number of different constructional features in the velocity type. Briefly, it may be understood that a pressure microphone is nondirectional, the velocity bidirectional, and a combination of the two are unidirectional. This should become clear in the following explanation.

Fig. 4-2 illustrates what happens to a pressure-type microphone when the diaphragm is excited by sound waves. In (A) the sound is originating in front of the face of the instrument and as the wavefront passes over the diaphragm it is brought from rest into the position shown by the dotted line. This results from the pressure of the condensation of air and is the principle of a pressure-type microphone. In (B) the sound is coming from the back of the mike, but it may be seen that the pressure of the wavefront causes the diaphragm to move inward just as in the case of (A). In (C) the sound is originating at the side, where the point of sound pressure causes the diaphragm to move inward just as in the other two cases. Hence, it may be seen that the diaphragm always moves in the same direction regardless

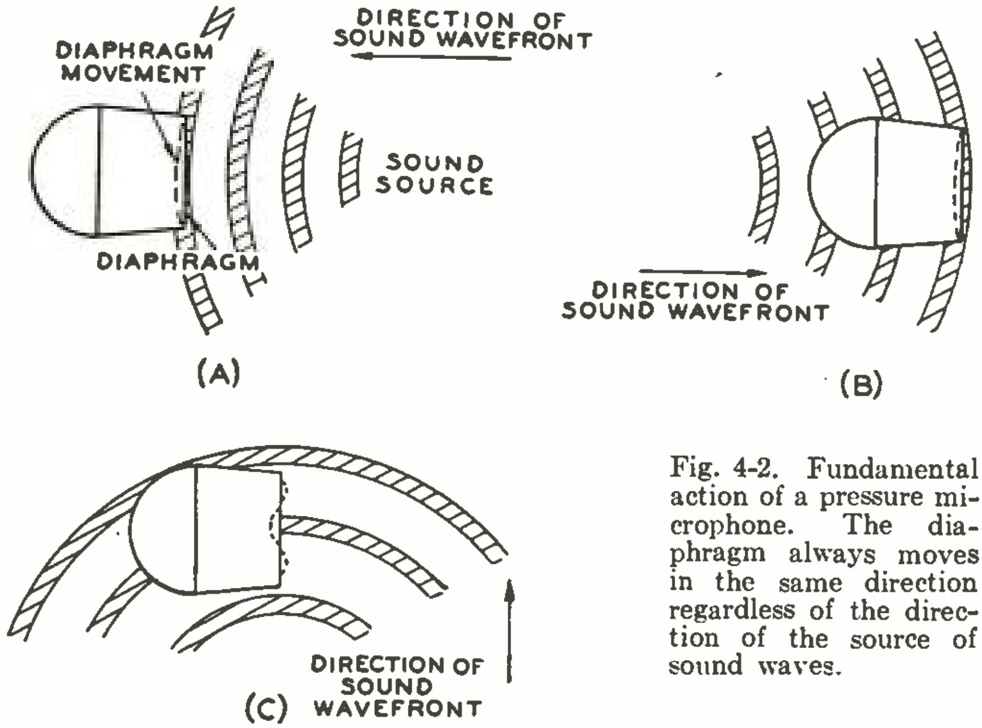


Fig. 4-2. Fundamental action of a pressure microphone. The diaphragm always moves in the same direction regardless of the direction of the source of sound waves.

of the initial direction of the traveling wavefront, and results in a nondirectional response pattern. Of course, it should be understood that this type of microphone can be made semidirectional by orientation of the case and by use of baffles. Fig. 4-3 illustrates this principle. This instrument, however, is not a truly directional microphone as are other types as will now be shown.

Fig. 4-4 illustrates the principle of the action of a velocity microphone when the ribbon is actuated by sound waves. It may be observed that sound waves on either face of the instrument (0° or 180°) will cause the ribbon to move although in opposite directions for a given wavefront. In Fig. 4-4(C) it is seen that sound coming from the side (90° to axis of face) will cause an equal and opposing pressure on each side of the ribbon, resulting in a theoretically zero movement of the ribbon, hence zero response to the sound. This results essentially in the "figure eight" or bidirectional response similar to Fig. 4-1(B).

The unidirectional microphone is just what its name implies; an instrument that by design is "live" on only one face, and "dead" toward the sides and rear. It is a very useful characteristic in broadcasting and recording studios.

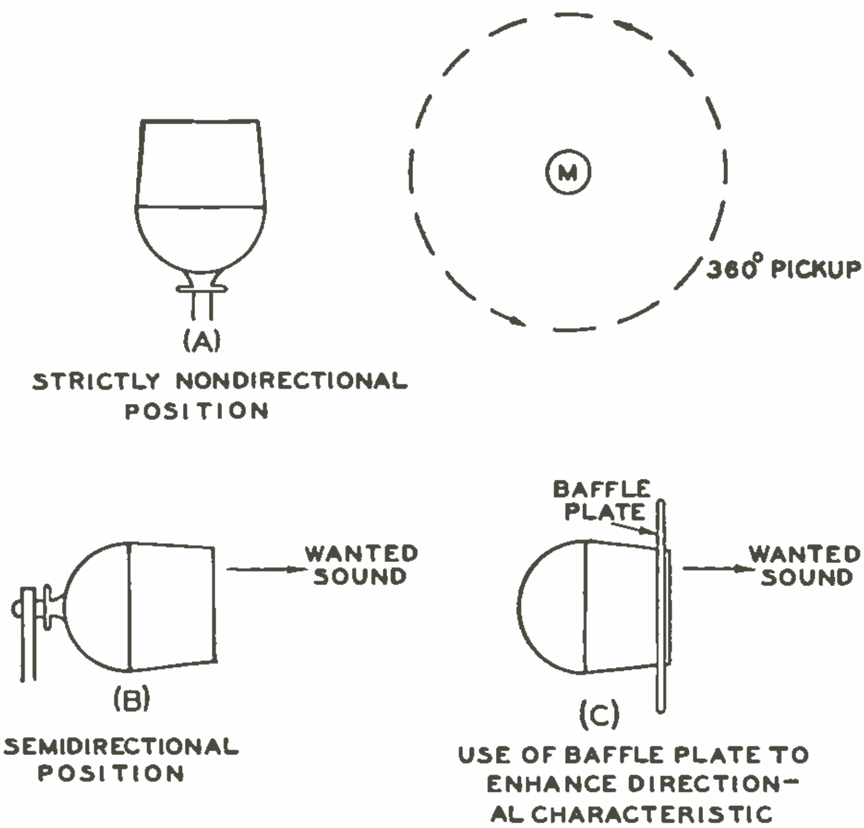


Fig. 4-3. The orientation of a nondirectional microphone for various types of response. The semidirectional orientations apply only for relatively high-frequency sound waves.

The unidirectional pattern may be obtained in a number of ways, but the first method used was the combination pressure and pressure-gradient principle, still widely used today. Primarily this consists of a pressure element, usually a moving coil, and a ribbon element connected in series. Their wires are poled so that outputs of the separate elements reinforce each other for sound coming from a given direction relative to the face of the instrument, and cancel each other for sounds coming from sides and rear of the zero-degree axis. This will be made clear by a brief review of the action of the pressure and pressure-gradient elements.

Looking again at Fig. 4-2(A) and comparing with Fig. 4-4(A) it will be noted that the diaphragm and its associated coil assembly (if an inductor mike), and the ribbon in the velocity microphone will respond in the same direction to the traveling wavefront. Therefore, if these two elements were connected in series and their connections properly poled with their respective magnetic fields, the output of the microphone would result in large amplitude to sound from this

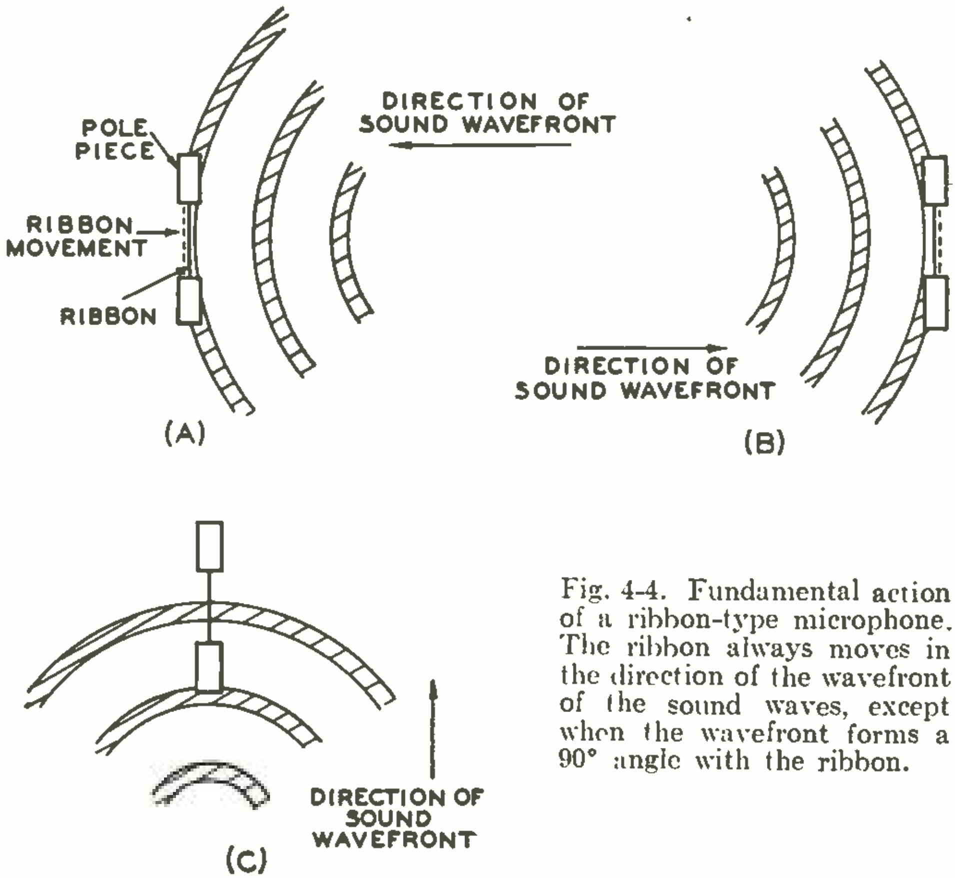


Fig. 4-4. Fundamental action of a ribbon-type microphone. The ribbon always moves in the direction of the wavefront of the sound waves, except when the wavefront forms a 90° angle with the ribbon.

direction. Now compare Figs. 4-2(B) and 4-4(B). It is observed that the pressure element will move in the same direction as before, whereas the ribbon element will move *in the opposite direction.* With the same wire connections as before, the voltage generated by the ribbon will oppose that of the moving coil. It is obvious, then, that sound emanating from this direction will be effectually canceled out at the microphone transformer, resulting in zero response. It may also be visualized by the reader that sound from the side will result in zero response from the ribbon element, but will still actuate (to a modified extent) the pressure element. This is the reason why the typical unidirectional microphone has a much wider angle of pickup to the front compared with the velocity (bidirectional) microphone.

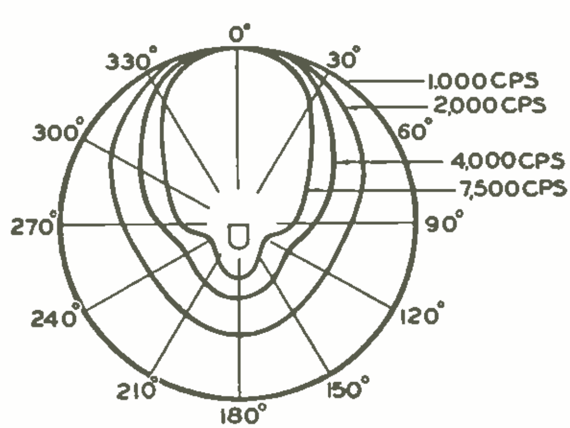
Effect of Angle on Frequency Response

Although in the hypothetical instance of the "ideal" case mentioned earlier, the microphone should be uniformly directional at all frequencies, such is not the case in practice. High frequencies travel in comparative beams compared to the low frequencies of longer wavelength, and a number of compromises of design and construction must be made. The use of a nondirectional microphone, for example, oriented so as to become semidirectional as in Fig. 4-3, holds only for the higher frequencies which are sufficiently deflected by the dimensions of the case. At frequencies of lower rate, approximately 1,000 cps or less, the microphone is still just as nondirectional in any position. The usefulness of such orientation is highest when used to reduce high-frequency feedback in public-address installations associated with a broadcast requiring a nondirectional mike.

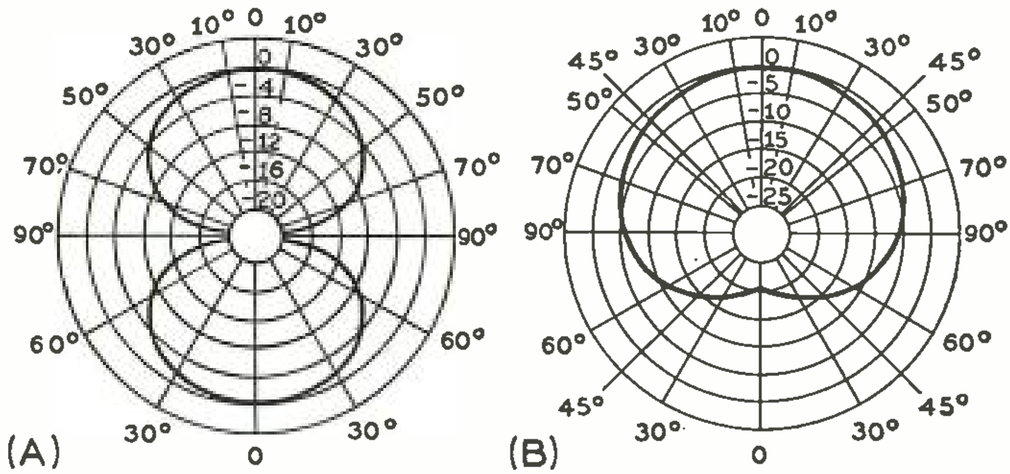
It is well known that high-frequency response falls off with increasing angles from the zero axis, or from line drawn through the exact center of the diaphragm or face. Looking again to Fig. 4-2(C), it may be seen why, whenever the wavelength of the sound wave from an angle is short compared to the width of the diaphragm, several points of unequal pressure exist causing an irregular movement. Thus it becomes obvious that for highest fidelity of high-frequency pickup at angles from the face, the diaphragm would have to be made smaller than the wavelength of the highest frequency to be considered in its use. In practice, this is impractical due to the extreme ly low output that would be obtained from such an instrument, and a compromise must be made. It will be noted, therefore, that all microphones are

Fig. 4-5. Variation of directivity with frequency for any type microphone. At low frequencies the pickup is nondirectional regardless of the orientation of the microphone case. The pattern is increasingly directional as the frequency increases.

Courtesy Electro-Voice, Inc.



directional at high frequencies, and have a wider angle of equal response areas at the middle and lower frequencies. A typical frequency-directivity response pattern is shown in Fig. 4-5.



Courtesy RCA

Fig. 4-6. The pickup response pattern of the ribbon or velocity microphone (A) is bidirectional and that of the combination ribbon and pressure type (B) is unidirectional.

In determining the proper use and placement of microphones for any given setup, it is important that the operator becomes familiar with the pickup patterns of the microphones used. These patterns illustrate completely the function as to amplitude and frequency response for varying degrees of placement about the face of the microphone. Fig. 4-6(A) shows the pattern of the RCA 44-BX velocity microphone, and Fig. 4-6(B) is the pattern of an RCA 77-B combination ribbon and pressure type instrument. There are several important points of interest relating to these patterns which show great differences in characteristics aside from the most apparent one, that of bidirectional and unidirectional pickup.

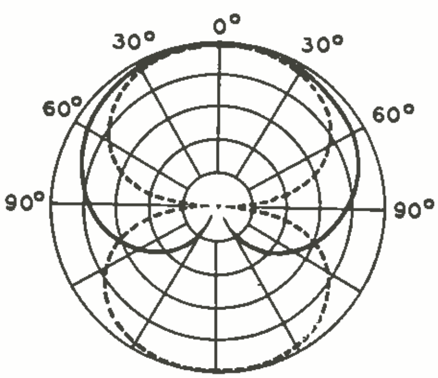


Fig. 4-7. The amplitude response of the 77B combination type microphone (solid curve) has a wider range than the 44BX velocity microphone (dotted curve). Such patterns are useful for making setups and eliminating unwanted sounds.

An analysis of the patterns reveals a much wider range of amplitude response for the combination pressure gradient (ribbon) and pressure type microphone [Fig. 4-6(B)] than for the ribbon type alone. See Fig. 4-7. Take for example the 1,000-cps curve for the 44-BX velocity microphone. It is noted from Fig. 4-6(A) that at an angle of 70°, the amplitude response is down about 10 db in respect to its response at a given distance at 0°. Now note on the 1,000-cps response curve of the 77-B combination type in Fig. 4-6(B) that the amplitude response at 70° is down only approximately 3 db from 0° reference. These patterns are useful for determining the setups necessary for discriminating against unwanted sources of sound, and for obtaining a particular relation between sounds of different sources. It can be seen that as a performer is moved around the microphone, loss of sensitivity may be compensated for by moving closer to the instrument.

It is clear then that characteristics of the type or types of microphone in use should be thoroughly understood. Fig. 4-7 is presented as a basic principle in using patterns of a unidirectional microphone of the combination ribbon and pressure type. It is a well-known fact that, because of the pressure gradient characteristic of the ribbon microphone, the instrument will favor the lower frequencies of longer wavelength under close talking conditions. For this reason announcers on such microphones must be at least 1.5 to 2 feet from the microphone. When close talking becomes necessary, however, the combination type instrument may be utilized by the engineer, who can then safely instruct the announcer to approach an angle of 90° with the

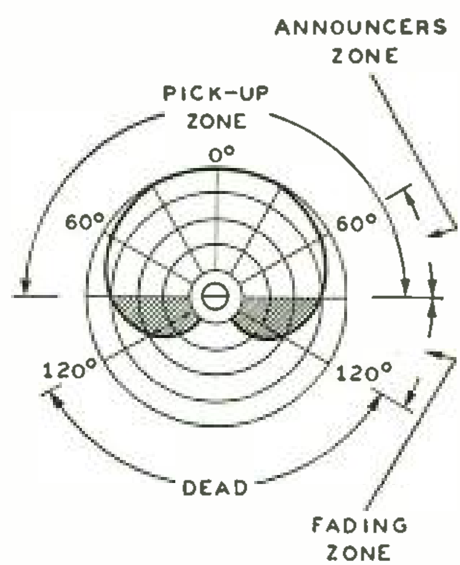


Fig. 4-8. When close talking into the combination type microphone is necessary, it can be approached as closely as desired in the zone between 60 and 90°. The lower response of the "fading" zone between 90 and 120°, does not affect the quality. This technique holds true for the combination ribbon-pressure element microphones, such as the Western Electric 639A and 639B types. Unidirectional mikes using a ribbon element only in conjunction with an acoustical labyrinth (such as the RCA 77-D) will obviously not react this way to sound angularly displaced from the axis.

Courtesy Western Electric Co.

face of the microphone as shown in Fig. 4-8, and work as closely as desired. In this position the ribbon element will contribute practically no energy to the output, leaving the pickup to the pressure element, which is not affected by the spherical character of close talking sound waves.

It may be seen that the "fading zone," where sensitivity falls off rapidly for increasing angles, is just as useful as the ordinary pickup zone, since the quality is just as good and a fine degree of shading may be realized by understanding its proper use.

As will be described and illustrated in the technical explanation of microphones in Part 6, a number of modern ribbon and combination microphones have an associated equalizing feature known as a "speech strap." In the "speech" position, close talking into the ribbon element will not result in excessive bass response. When the same microphone, however, is used both for the musical pickup and the announcer, the strap is placed in the "music" position, and the announcer must work the mike as explained above.

Facts or Fancies?

"The microphone is a mechanical extension of the human ear." This is purely a fancy! The fault in this definition lies not so much in the literal meaning, but in the implications that are involved. If it were true, even if we only have one ear, we could walk into a studio or on the stage and place the microphone at the spot where the one good ear could "hear" the orchestra, the soloist, the chorus, and the announcer. It won't work. Yet this conception is probably the contributing factor in the reasoning of an operator who "sticks up" a mike and proceeds to handle the show by "riding gain." Of course, there is the more ambitious type who spots a microphone for each section of the orchestra, then one for the soloist and announcer, and several for the chorus. Granted that there are some cases (particularly as applied in remote controls as explained in Part 3) where this is necessary, it will be made clear in discussions to follow that this condition is the exception rather than the rule, for regular broadcast studios.

Why is it that a microphone cannot be treated as an extension of the human ear? To understand this it is necessary to review a fundamental theory that is an old story, but absolutely essential to understanding the mike from an operational point of view.

Unlike the human hearing system which is binaural (two-eared) the microphone is monaural (one-eared). Please bear in mind that this is true regardless of the number of microphones used. If we are using one mike or ten, the sound is collected into one channel and received through one loudspeaker. Physically the difference is that in a monaural system the sense of *direction* is lost, while reverberation is somewhat more noticeable, making the apparent distance of the sound source seem greater than when actually listening binaurally. Any operator who has ever set up a microphone in a particularly live hall has experienced this phenomena. He is able to hear quite clearly the conversation of the two people somewhere in the hall when listening with his two ears, but when listening with the headphones through the amplifier the sound seems much more distant and extraneous noise is very high. This brings up the all important psychological factor closely linked with the above physical one.

This psychological factor is the unconscious ability of *focusing power* when listening binaurally, or even with one ear plugged as much as is physically possible. The ear, associated as it is with the nervous system and the brain, tends to exclude extraneous noise that is present and "focus attention" on particular sounds. The microphone is a mechanical device not associated with any means of concentration except as it may be deliberately used to achieve the desired result in the listening ear at home.

A practical example is evident to all who have ever dined in a public place that, for instance, may have a salon orchestra rendering dinner music. Our table may have been out in the center of the room, and although the usual clatter of dishes, hum of conversation, and all the various noises are quite noticeable, the music provided may be found very enjoyable when we care to listen. Yet it takes little imagination to realize what the result would be if a microphone were placed at this same table. Very little of the music would be heard, the few strains coming through would sound far, far away. The pickup would be nothing but a hopeless hodge-podge of voice, noise, and confusion.

First, then, let's do away with the misleading idea that the microphone is a mechanical extension of the human ear. *Do this whether thinking of a.m. or f.m.* Think rather of the microphone as an instrument *that must be used* according to its possibilities and limitations to *achieve* an exact replica *in* the listening ear of the original program content.

Another point needs mention:

"Microphone technique is different for f.m. than for a.m." Well, this is still a somewhat controversial subject. The author feels, however, that the following analysis should be carefully considered before taking the statement literally.

It is obvious that the principal difference between a.m. and f.m. is in the means of r-f modulation, transmission, reception, and demodulation. Many a-m stations today have an audio system that is the same as the audio system of the best f-m station. Now consider only the job of the control-room operator or production man responsible for mike setups. He will monitor the program from the loudspeaker in Control or Production, whether or not the audio signal is to feed an a-m or f-m transmitter. It's a foregone conclusion that the program will sound no better in the receiver than it does in Control, regardless of the means of transmission. It is, of course, true that the f-m listeners will hear more nearly what is heard in the control room than will the a-m listener. It appears then that the rule should be, "Microphone technique is *somewhat more important* for f.m. than for a.m." The author feels that many readers who are still just contemplating f.m. can begin right now to use "high-fidelity microphone technique" in a-m studios to prepare for f.m. In all likelihood the critical a-m listener will notice a vast improvement in program quality.

Patterns of Response

Insofar as program setups are concerned, the foremost characteristic of a microphone is its pattern of response. There are several very important factors in using a microphone pattern to obtain a desired result.

The basic point to keep in mind is that a response pattern as illustrated for any particular microphone *is plotted in a perfectly "dead" room to avoid any practical amount of reflection of sound waves.*. In other words, a pattern of response will hold true when not influenced by any enclosure, such as when used outdoors. It becomes obvious here that the manner in which a microphone response pattern is affected by the acoustical nature of the surroundings becomes an interesting and highly important item.

In general, it should be understood that the pattern will hold more nearly true in a "dead" studio, and be altered more and more as the surroundings are made more "live." Let's take a hypothetical case of a setup involving two sound sources and a bidirectional microphone.

Fig. 4-9(A) illustrates the theoretically dead room and "ideal" curve of a bidirectional microphone. Sound sources *A* and *B* are of the same intensity and same distance from the mike, but *B* is directly on the zero response axis. Since sound source *B* will excite both sides of the ribbon equally, no movement of the ribbon will result from this source, and no output voltage will be created in the microphone. Therefore, the total output voltage will consist only of the impulses received from *A*. Thus the pattern holds true.

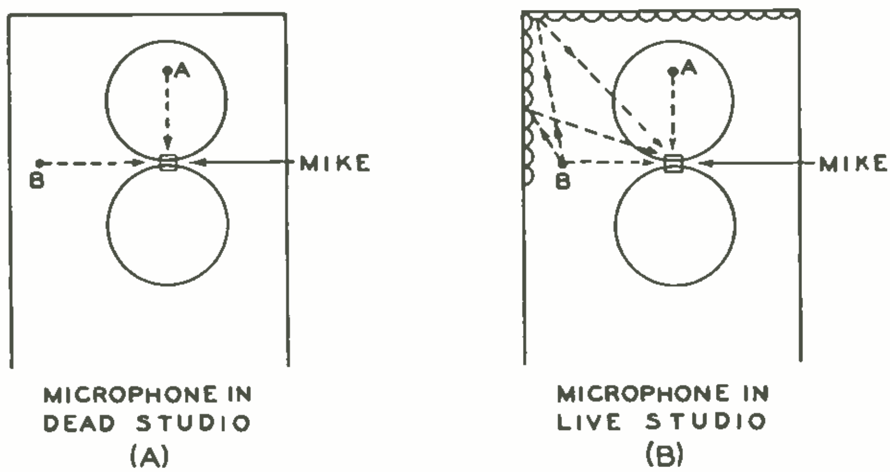


Fig. 4-9. Response patterns for a directional microphone under ideal conditions (A), and in a studio where there is a great deal of reflection of sound waves from the walls and surroundings (B).

Fig. 4-9(B) shows the same setup in a "live" room. It is still true that *B* will excite both sides of the ribbon equally resulting in no response to the direct wave, but now we have reflections. Sound will be reflected from the walls back into the sensitive side of the mike, and although reduced in intensity, will add to the sound from source *A*. The only difference now between the two sound sources is not only one of intensity, but ratio of reflected to direct sound. Naturally, this ratio is greater for sound source *B* than for source *A*.

It is, of course, obvious that there is no such thing as a perfectly dead studio or hall. This example, therefore, is rather crude but serves to illustrate the basic idea of how acoustical treatment influences the polar response pattern of a microphone. This understanding is imperative from an operational point of view.

From this and the previous discussion we have four fundamental operating points for the microphone.

1. As a sound source is moved about the mike, loss of sensitivity may be compensated for (if desired) by moving closer to the instrument.

2. The ratio of reflected to direct sound may be raised in sufficiently

live surroundings by using greater angles from the zero axis of the mike. (Especially true of bidirectional microphones.)

3. The more live the acoustics, the greater will be the effect.

4. High-frequency sound sources must be more nearly "on beam" for a given distance to achieve the same intensity of response as lower-frequency sound sources.

What about Acoustics?

The nature of the acoustical enclosure about the microphone is the outstanding factor that has prevented any establishment of definite standards in microphone setups. If there were such a thing for example, as a "standard studio," designed to approach as nearly as practicable the ideal condition of sound dispersion, the setup for any particular musical organization would be a simple matter anywhere when once worked out. Naturally, such is not the case. It is probably safe to say that there are no two studios or auditoriums anywhere in the world that are acoustically alike.

Right here, the author feels it advisable to bring out a point so far neglected in the scanty literature on mike technique. This point is that there are, and will be, a great number of operators and producers who will *not* have adequately designed and acoustically controlled studios in which to work. This is the most deplorable situation existing in broadcasting or recording, but outside the influence of the average operator concerned with this study. All we can do is hope, and then plead, plead, plead, and plead some more until the owners become cognizant of the extreme importance of a modern studio. It is a sad state of affairs that even some of the most recent f-m broadcast installations, building completely new studios independent of any a-m company, are neglecting this feature. Most of the good articles appearing thus far on microphone setups have been concerned with well-designed, musically "live" studios of the network centers or more production conscious independent owners. Yet how can a study of this kind be practical to the average operator wanting to do his best in studios of the same design as those of twenty years ago? We want this text to be a practical one, and will attempt to show some definite rules that may be adopted to meet any acoustical condition, even though it should not be necessary to do so. "High-fidelity technique" used in a good modern studio is not that kind of technique for the type of studio with which many readers are concerned. We will have to plant our feet squarely on the ground and face the situation as it really exists.

The so-called "modern studio" may not yet be the ultimate in design, but it is certainly a far cry from the previous rectangular-shaped, acoustically deadened rooms called "studios." The walls remain "live" to musical sounds but are broken up acoustically in some manner to avoid standing waves while still achieving a maximum response to diffused, polyphased high-frequency sound. Good tonal brilliance is obtained with a minimum of microphone and control-board manipulation. Recording studios seem to have kept more abreast of recent acoustical development than have broadcasters outside of the network centers.

Summary of Fundamentals

Let's take the basic factors thus far described and correlate them into a well-rounded conception of what we have to work with.

Firstly, there is the monaural response of audio transmission. We must expect to lose the sense of direction of the various sound sources until, if, and when stereophonic transmission becomes a reality. Sound perspective, then, must come from *relative distances* of the sound sources. A sound will still appear "close" or "distant" in a monaural system. Also due to this type of response, "focusing power" must be achieved by orientation, rather than the subconscious and automatic means of the human hearing sense.

Secondly, strive in any microphone setup to achieve the best results possible as determined by the sound from the monitor speaker. Do not be confused by supposed differences in a.m., f.m., recording, etc.

Thirdly, bear in mind the microphone response pattern and select one suitable for a particular purpose as explained later. Remember the influence of the acoustical treatment surrounding the pickup area.

And fourthly, we realize that many do not have available the latest in studio design so necessary for brilliance in musical tones. We will attempt to help overcome this difficulty by showing what operational modifications are necessary in the older-type general-purpose studios to achieve best results possible.

Microphone Phasing

It is very important in any sound application using two or more microphones fairly close together, to have them properly phased in output. If this is not done the outputs will oppose one another, reducing the total signal and introducing some distortion. This is the same thing that happens in multiple loudspeaker systems when the voice coils are out of phase, causing a "pumping" or bucking of sound at points in the overlapping field of the speakers.

The best way to phase two or more mikes is to hold two in the hand at the same time while talking. Turn one mike on and note the volume-indicator reading. Be sure not to alter the volume of the voice when turning on the second mike with the same volume-control setting. If the volume now decreases in level, the connection of one of the mike cables at the amplifier input terminals should be reversed. Each additional microphone should be checked in this manner against the first microphone. Phasing may be accomplished by using a patch-cord between any two terminations of the circuit on the jack panel, and reversing *one* end at a time during the test.

The Announcer

The single announcer is the logical starting point for our discussion of mike technique. It must be understood here that we are not concerned at present with announcing over a background of music or any form of dramatic presentation.

"What is there to discuss then? He just steps up to the mike, or sits down in front of one—and talks!"

You're right. He does. But the reality of an error does not necessarily justify the error.

Announcing alone occupies a considerable portion of the broadcaster's schedule. Correct voice transmission is so very important to radio simply because we do not have the sense of sight to help our impressions; the voice is the complete medium of expression. The intake of the breath, the most subtle inflections, the style of delivery, the original voice *timbre,* all are factors in *naturalness* of the voice. Any or all of them may be severely affected by the announcer's relation to the mike. All of us in the technical end of radio must at some time in our career realize that our engineering training has not encouraged a feeling of artistic values or sense of showmanship. We are very apt to lose sight of the real reasons behind the keys, faders, and perfectly matched impedance of the control system. They are designed this way so that the electrical impulses may correspond to the thunderous crescendos of Wagner's "Die Walkure" or the light delicate strings of Debussy's *"Festivals."* The moods of music! The moods of voice! With these intangible qualities of human experience are the wires and switches and dials and knobs of the technical department concerned. If we grow ever more conscious of this in the course of our work, we will find whole new vistas of sound and sound control opening at our fingertips.

Here is the first thing to do with the announcer.

Get the microphone away from his face.

Mugging a mike is a terribly deep rut and strong habit to get out of. We will likely meet resistance from the announcer himself. In many cases we literally cannot do it. Some announce desks have microphones permanently mounted on them, and the announcer cannot move back without holding his script uncomfortably in his hand with no arm rest. This is plainly and simply an engineering error. Like fingerprints, there are no two voices exactly alike. For every voice there is a definite relationship with the microphone which allows the most natural and pleasing reproduction for that particular combination of voice and mike.

For voice work alone, *there is no distance less than two feet from the face of the microphone that will assure "naturalness" of that voice!* A distance of two feet, incidentally, should only be used for the "softest" voices. Compare this with your own observations of studio operating practice. What is the distance most announcers use? Probably somewhere between four inches and a foot. Voice waves at this distance from the mouth and throat cavities do not create the electrical impulses that correspond to the natural character of that particular voice. Setting the mike on "voice" adjustment, where such is incorporated in the mike, helps in cases where it is absolutely necessary to work very close, (to cover up background for example) but it is *not* natural transmission.

Do this as an experiment if it is possible: On rehearsal, set up microphones (in addition to the regular "announce mike") three, six, and nine feet away. Stagger them enough so that no mike will be in front of the other. If the regular microphone is immediately in front of the announcer's face, we will have to move it lower or to one side. Don't tell him what is occurring; we are just "testing mikes." If he is told to start reading at a great distance from one microphone and then to move closer, he will subconsciously alter his volume and we will not get a natural check.

Now turn only one microphone on at a time. Start with the farthest mike. Unless you are so conditioned to hearing the "mugging voice" that you can accept no other, you will find a new experience in naturalness of voice transmission. A distance will be found where the voice begins to sound "hollow" in a live studio, or "thin" in a dead one. Use the distance just a shade closer than this point.

In small "announce booths" where space is at a premium and where distance would create the "barrel effect" due to the proximity of the studio wall to the microphone, the mike should be suspended over the announcer's head at a distance of several feet. We are borrowing this technique from television where the mike must be kept out of the range of the camera and is often as much as fifteen feet from a performer. It should be borne in mind, however, that the sound in television is only a secondary expression to the picture. In other words, all of the intent and meaning need not be embraced in sound alone as in regular broadcasting.

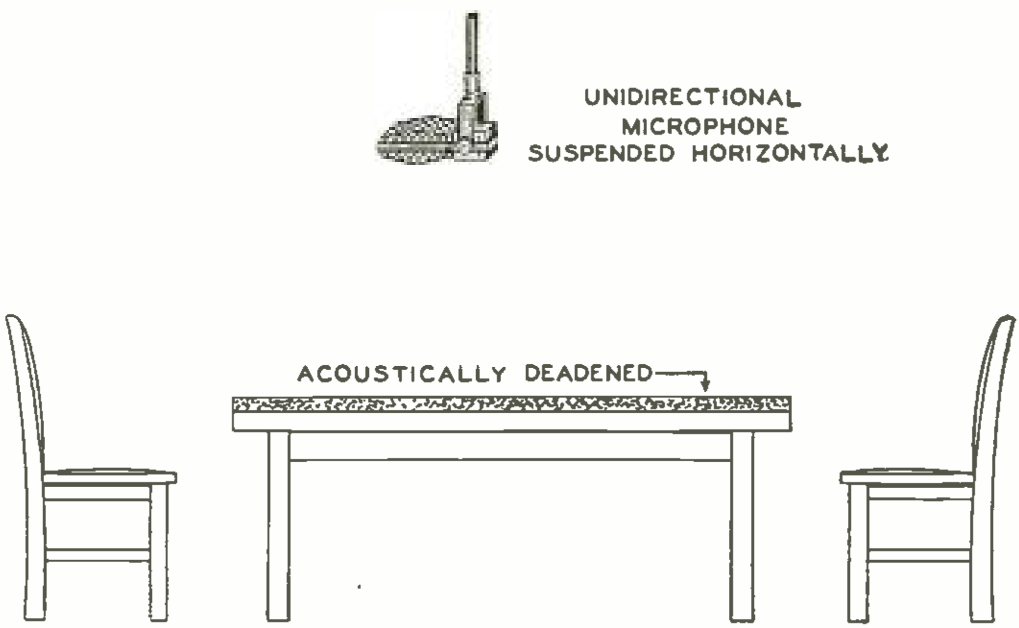


Fig. 4-10. The use of a suspended mike for interview-type setups where two people are seated before an acoustically-deadened table.

This same technique is most convenient when two persons are seated at a table and using a single mike as illustrated in Fig. 4-10. *The table top should be deadened* to sound by acoustical treatment or a heavy cloth cover to prevent reflected sounds entering the mike. This is very important when using the suspended-mike technique.

How Many Microphones?

In an approach to the study of specific types of program setups, the question of the number of mikes that should be used will invariably arise, especially on a large show. Always keep in mind the previous discussion on monaural character of the system and the inability of focusing power unless deliberately used as a means of concentrating the attention.

The greatest weakness in mike setups for large shows has always been the use of too many microphones. There is bound to be some distortion (however slight) in multiple mike setups due to time lag of sound waves creating phase additions and subtractions at the various positions of pickup. This source of distortion, however, is only minor compared to the other faults of this technique. Aside from the operational difficulties of handling a large number of channels on the mixing panel with greatly increased chances of error, the control man and his board have now taken the place of the conductor. All of the dramatic interest, the emotional pattern written into the original score, plus even the conductor's interpretation. is now placed in the ratio of fader adjustments, the reaction and psychological temperament of the operator; in other words, in too many variables.

Let's establish a foundation upon which we may build a workable structure for determining the correct number of microphones for a given show. Let's also be practical and realize that a great number of operators do not have the very latest studios and hours of rehearsal time on one program.

1. Whenever it is physically possible in your alloted time of setup, so arrange your performers about a *single microphone* (following suggestions given later for each type of show) so that the over-all balance is correct. If you have the time to do this, you achieve "balance" in the proper manner by positioning rather than mixing various sources on the control board.

2. Assuming now that your time is running out and you are still having trouble with a particular section in obtaining balance. This is more apt to occur in a dead studio than in an acoustically correct live studio. Certainly your present use of one mike is not achieving maximum fidelity, and another mike on the troublesome section will have to be used. Many times in practice it will be found that the second pickup need be used only at times and not all through the show (such as rhythm section accentuation of an orchestra at a few spots in the score).

3. Some of the more complex shows will positively require more than one microphone for proper pickup. Take, for example, a variety show consisting of a drama cast, a chorus, and an orchestra. Remember that the microphone *does not* "focus attention" as your ears would

do in the studio. Of course, it is likely that a program of this type will originate in the larger stations having modern studios and plenty of rehearsal time with the operator. You will find that the more time you have, the less number of microphones necessary up to the point where you have reached the absolute minimum.

4. The rule, then, is simply this: *Use the absolute minimum number of microphones for a given program and set of conditions surrounding rehearsal time.*

So much for the basic theory of microphone technique. Let's go on with more specific program setups.

The Piano Pickup

The single piano pickup is the simplest setup we have in any kind of studio since it is perhaps the least affected by acoustical nature of the room. But there are "tricks of the trade" even in the simplest programs.

First, regardless of how softly the pianist plays (so long as he is unaccompanied by other instruments) the microphone should *not* be placed up under the lid. Most of you who have had considerable experience with using the mike have known the amount of distortion arising from the close proximity of large physical objects to the microphone. It should then be obvious that a pickup under the lid and close to the sounding board will not allow natural transmission of the piano tones. "Close miking" of this kind to almost any instrument will cause many "peaks" on the vu indicator that are inaudible, with a consequent necessity of holding the volume down with great loss of musical brilliance. It is true that many operators have become so accustomed to this type of piano setup that they are in the same position as those in a habit of hearing the close-talking announcer. It may be a "familiar" sound, *but it is not natural reproduction.*

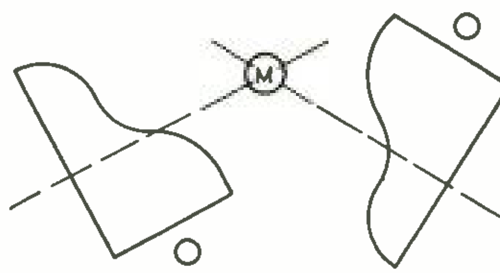
Remember that it is the listeners' prerogative to operate his receiver tone controls to suit his particular set, acoustical conditions and hearing sense; it is the business of the broadcaster to transmit the natural sound of the original content.

Here is the best method of determining the setup for a single piano. Start with a distance of 20 feet, head high. In a dead studio, this distance will probably result in a "thin" response, especially on low passages. In live studios, the sound may be too reverberant for clear-cut transmission. Move the microphone in on a line drawn through the center of the sounding board until the tones are full bodied and just the right amount of reverberation exists. This distance is seldom less than eight feet, and will allow tonal brilliance and balance between lows and highs that is sacrificed in close-up technique.

The piano should be placed on full stick.

Now comes the final check for balance between bass and treble, in other words, the check of the player's left- and right-hand pressures. If the pianist happens to use approximately the same pressures on both hands, the above procedure is usually all that is necessary. But assume he has a heavy right hand, and bass response is somewhat weak in relation to the highs. Keeping the mike at the same distance, move it from the line down through the center of the sounding board down toward the tail of the piano. This method increases the response from the bass strings. If the pianist should have a heavy left hand causing loss of highs (this lack of highs is also apt to occur in dead studios), the mike should be moved toward a line drawn through the *hammer line* of the piano, or even over to an imaginary extension of the *keyboard*. This will increase response to the treble strings and decrease that from the bass strings.

Fig. 4-11. A microphone *M* for a two-piano pickup is placed between and slightly higher than the pianos.



The twin piano team imposes only slight additional requirements. Fig. 4-11 illustrates the most satisfactory orientation of the two pianos with the microphone. The temperament of the pianists must, of course, be considered and the lead piano given prominence by moving the mike closer to that piano if the accompanist is "heavier fisted." The above procedure for bass and treble balance must be followed not by moving the microphone, but the piano itself.

The Vocalist and Piano

There are three variables here; the distance of the vocalist from the mike, the distance of the mike from the piano, and the distance of the vocalist from the piano. There is no excuse for using more than one microphone unless the time allowed for rehearsal is zero.

Here is rule number one. Listen to the vocalist in the studio first. Does he or she "sing out" with the chest muscles, using a large volume and dynamic range? Or is the vocalist the "crooner" type, using only the larynx and throat for emphasis? You will find every vocalist in one or the other category, generally speaking.

Now excluding the piano accompaniment, we know whether the vocalist must be near or far from the microphone. One who "sings out" with his full volume range will be placed anywhere from 12 to 6 feet from a microphone when a piano alone is used as background. On rehearsal, always start with the greatest distance. The goal we are striving for here is to be able to adequately enjoy the lowest volume and the highest volume *without riding gain on the fader control.* This part of the "balance" can *always* be achieved by careful rehearsal checks.

The balance between vocalist and piano accompaniment is not always so simple. A good pianist or one familiar with a particular vocalist's style will automatically adjust his volume to the *pianissimo* and *crescendo* of the singer. The mike may be placed about 8 feet from the piano with the vocalist on the opposite side (calling for a bidirectional microphone) at the distance determined by trial. It is well to point out here that a very common error of operators is in taking a "vocal solo" too literally. The "presence" of the piano must always be there, with only slight emphasis on the voice. It should be a *blend,* with, of course, the voice always a little predominant. This does not permit the so often noticeable weak background of piano tones.

When the pianist insists on playing so loud that his accompaniment smothers the vocalist (some pianists cannot alter their volume and still play well), the dead side of the vocalist's microphone must be turned toward the piano. This is practically always the only adjustment necessary from the 8-foot distance between mike and piano. If the studio is very live and the piano tones are still too predominant, turn the piano around so that the lid opens toward a wall of the studio and deaden the wall with drapes. Don't, at any time, move the vocalist closer to the mike, where "riding gain" is necessary on the natural dynamic range of the voice.

Small Strings Setups.

By this we mean the small salon groups, string quartette, or hill-billy groups. Their music is intimate in style, requiring good instrumental definition. This calls for comparatively close mike setups. *But not too close.* The author is always hesitant about using the word "close" since the reader is apt to take it as meaning directly into the face of the instrument.

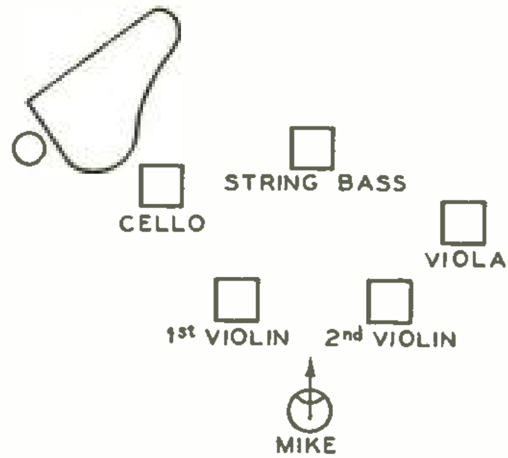


Fig. 4-12. Microphone setup for a salon orchestra or a small string group. By focusing the microphone properly all instruments will be picked up in their proper volume relationship.

Consider a small salon orchestra consisting (most popularly) of several violins, a viola, a cello, a string bass, and sometimes a piano. Due to their comparative volume they are usually placed in that order from the microphone. Fig. 4-12 illustrates the general orientation of such a group with the mike.

Now assume that the approximate distances are: violins 4 feet, viola 6 feet, cello 8 feet, and string bass 10 feet, with piano somewhat off mike at 10 feet. Also assume that the violins are too predominant for proper sectional balance and comparative "presence." What would you do first?

Here are the possibilities if you move the instruments: The violins could be moved farther back or to one side in a less sensitive zone or the other instruments could be moved closer in.

But have you remembered the "focusing-power" principle? Look at Fig. 4-13. You can achieve quite a range of sectional balance by the simple expedient of height and tilt adjustment. If the violins are too predominant, raise the mike and focus on the other instruments. If the violins are weak, the mike should be lowered and focused on the violins. This is a better method than moving the predominating instruments to one side in a less sensitive area of the mike, since the higher frequencies containing the overtones are very important for wide-range pickups.

There is little difference in pickup for this type of musical organization between dead and live studios, since the setup must be intimate in character, with high direct to reflected sound ratio, minimizing the effect of the acoustical nature of the studio.

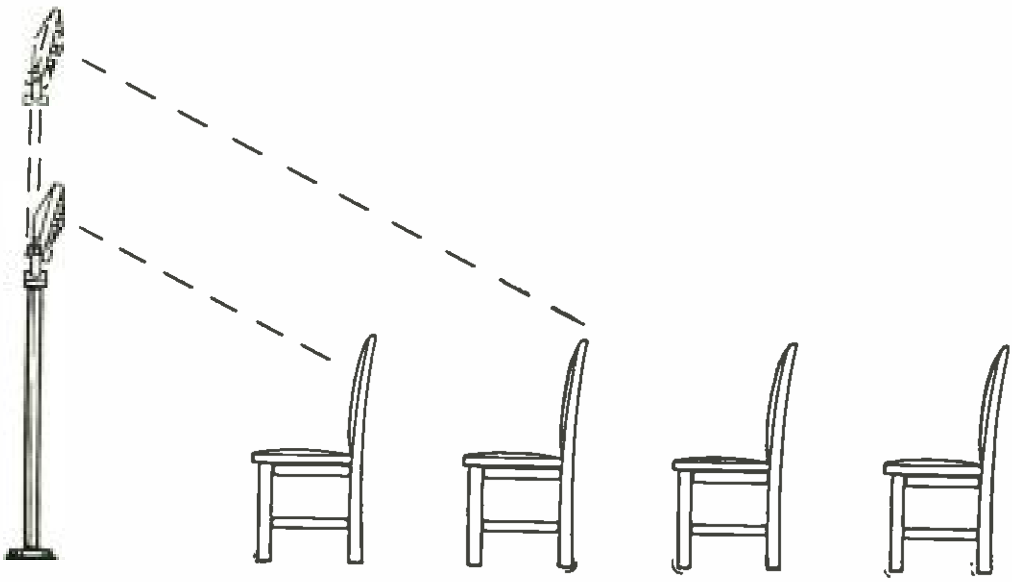


Fig. 4-13. A great range of sectional balance may be obtained by adjusting the microphone for height and tilt, as shown here.

Small Orchestra Setups

In this category come a large number of organizations presenting popular, serious, or variety music. They may consist of a combination of brass, strings, and reeds, and are anywhere from 4 to around 15 in number.

Again the first step is to visualize the instruments used in comparative power outputs as follows, (softest to loudest):

1. Violins. trumpets, or trombones (muted); guitar
2. Clarinets, saxophones, xylophone, vibraphone
3. String bass
4. Piano
5. Trombone (open belled), trumpets (open belled)
6. Traps and bass drums; guitar (electrically amplified).

The above are the most likely instruments to be encountered in such a setup. As before, the most likely approach is to arrange them in that order from the single microphone.

Look now to Fig. 4-14. This is a "live studio" practice, starting with the violins at about 8 feet. When initially checking balance of sections, remember the height and tilt adjustment for focusing power to obtain proper blend. Then, and not until then, try moving any troublesome section.

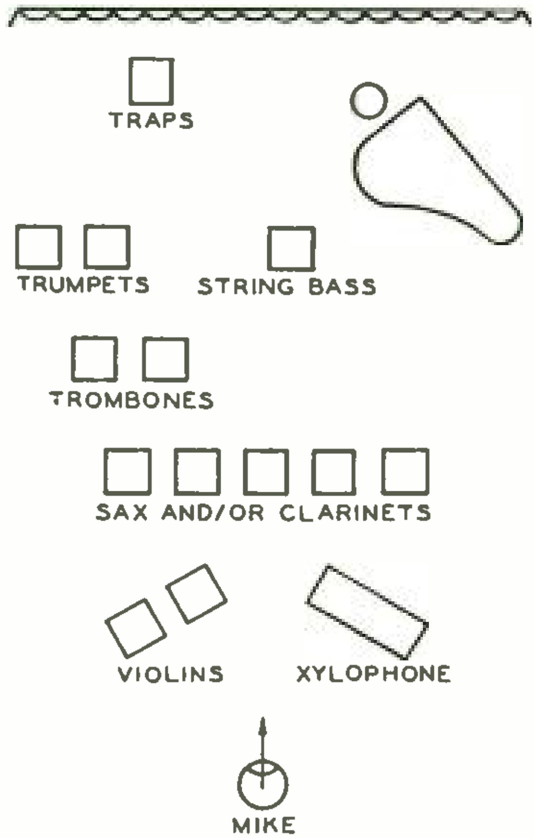


Fig. 4-14. Live studio pickup of small orchestra using single unidirectional microphone. Special consideration must be given those instruments which are "muted" (such as trumpets or other brass instruments), since they will obviously not be heard properly if they remain their normal distance from the microphone. If they can not be brought to the front of the group, a second microphone may be spotted near them and used only when the muted instrument is played.

When you think it becomes necessary to move instruments, keep these principles in mind:

A predominating section *might* be too loud not because the relative distance between instruments is incorrect, but *because the microphone is too close to the entire outfit*. Move it back.

A weak section *might* be too soft, not because it is too far from the mike, but because all instruments are too close to the microphone. *Move it back.*

In other words, the farther back a microphone is placed (within the limits of acoustically allowable distance), the better the chance of a good balance between all sections.

Another very important item is the treatment afforded muted trumpets or trombones. When their bells are muted, the instruments must be very close to the microphone. We really mean close here, about 2 feet. If the players cannot or will not step from their regular positions to one immediately in front of the mike, a separate microphone must be spotted just in front of that section. Obviously, it need only be used during the time when they *are* muted.

Now consider a studio of older design without the "liveness" of a modern studio. When the musicians number around 12 to 15, it is often difficult to get good sectional balance on the above setup with only one mike. Even though the farther instruments may contribute about the same number of volume units, their "presence" may be "thin" due to the lack of reinforcement of harmonics and overtones that occurs in the more modern studios. Also the mike must be a little closer in a dead studio, emphasizing the discrepancy in sectional "presence" because of the dwarfing of closer instruments on the farther instruments.

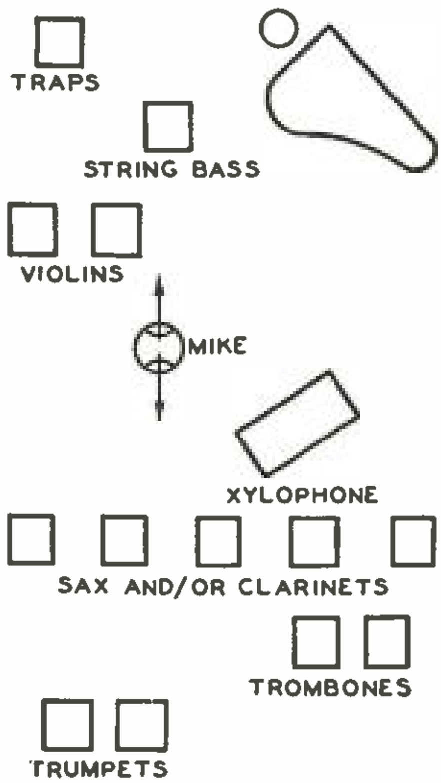


Fig. 4-15. Same small orchestra as in Fig. 4-14 but split, to use bidirectional microphone. This is effective in dead studios, or where space is at a premium. Ordinarily, dead studios produce a "thin" over-all orchestral sound due to the absence of a great number of harmonics absorbed by the walls. The use of a center microphone with a split orchestra brings the weaker orchestral and harmony instruments closer to the microphone to produce a richer orchestral sound.

In nearly every case, however, we can still use only one microphone by using the setup illustrated in Fig. 4-15. Here we are using a bidirectional ribbon and the instruments must be more nearly "on beam" due to the narrower angle of response in comparison to the unidirectional mike of Fig. 4-14, and due to the less live acoustical nature of the studio. More time, and movement of players will be necessary here. But it is far better than the usual procedure of using additional microphones.

Adding a Vocalist

A vocalist with the above musical type of program imposes special problems unless the organization is trained thoroughly in prior broadcast production technique.

The ideal arrangement would be for the vocalist to be placed in front of the orchestra facing the mike at several feet away depending upon the singer's style as previously discussed. This arrangement, however, is often not practical; as, for instance, when the mike is raised and slanted so as to obtain proper balance of instrumental tones. It also imposes restrictions on the actual musical arrangement of the accompaniment since, if brass is used instead of just strings and possibly reeds, the voice will be dwarfed out.

Where such occurs, there is no alternative but to use a second microphone for the vocalist, preferably a unidirectional type with dead side toward orchestra. This mike may also be used for the announcer.

The Symphony

The large symphony orchestra pickup is based upon exactly the same principles as heretofore discussed. We are, however, concerned with a combination of each type of orchestral setups into one grouping; strings, woodwinds, brasses, percussions; and quite often a choral accompaniment. The grouping of instruments and the type of musical score become much broader in scope. The most helpful characteristic here is the fact that such a program is less apt to be attempted in an inadequatedly designed studio than in the formerly discussed program.

It will be found that generally the usual physical arrangement of the orchestra for regular audience listening will be satisfactory for broadcast purposes. On the initial trial of the main orchestra microphone, several mikes should be suspended at the most likely pickup positions in order to avoid the confusion of moving mikes, ladders, and personnel. The general area for these microphones is 15 to 20 feet high and 15 to 25 feet in front of the violin section.

Again it may be necessary to point out that if a vocalist or instrumental solo is called for, it is almost always necessary to use a second mike to achieve proper sound perspective. *The main orchestra microphone will not act like the human ear "in person."* Remember, too, that vocal or instrumental solos are *not* to be entirely predominant; the orchestral accompaniment must be very much present. Always try, if possible, to get the conductor to listen to the control monitor for a final check on balance. If this is not possible, get some responsible member of the organization to pass on it. The very best control or production men do this simply because every symphony organization has an arrangement of score or possibly a distinct interpretation of the original score with which you cannot be expected to be familiar. This has, many times, necessitated a slight rearrangement of instruments in relation to the microphone.

Along this same line it has become evident that in spite of the usual superior results obtained with a single well-oriented microphone, it is often necessary to deviate from this practice for true "high-fidelity" pickup. The main orchestra microphone, back far enough to obtain the proper blend of all instrumental tones, will faithfully pick up the delicate, distant tonal beauty of the violin passages in *Clair de Lune.* Most music lovers, however, criticize this same microphone setup for such numbers as the Strauss waltzes, where the tonal perspective of the strings should be closer and more strident in quality. It all boils down to the fact that the microphone is *not* a mechanical extension of the human ear. You will find many leading conductors and producers of symphony broadcasts insisting on an added mike

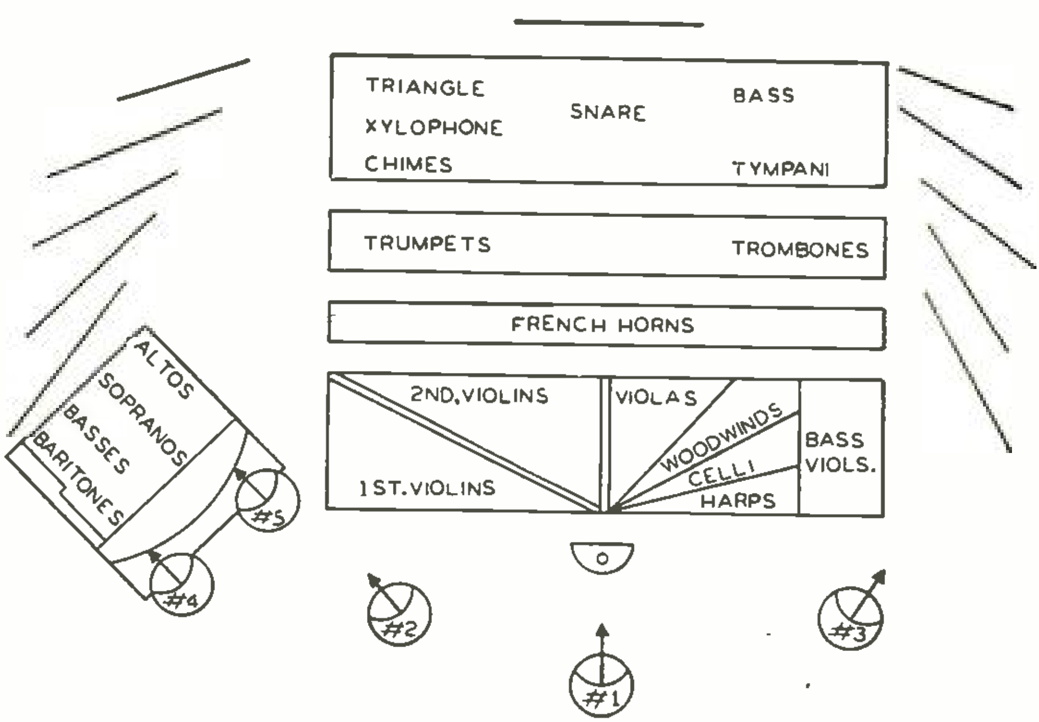


Fig. 4-16. Orchestra and choral arrangement for a "live" studio, mike 1 for main orchestra; mike 2, soloist; 3, announcer; 4 and 5 for choir, when clarity of diction is needed. Mike 2 used for over-all choral effect.

suspended directly over the strings or other sections of the setup to be turned up *only on cue.* This procedure supplies the missing psychological factor in obtaining microphone-to-sound perspective. When choirs are used with a symphony orchestra, it is also necessary to use a separate mike due to the necessary distance between the total combination and the necessary "focusing power" to be obtained by the use of the microphone. A typical symphony and choir arrangement is shown in Fig. 4-16.

Symphony Setup Preliminaries

The major difference from the beginning in considering a "setup" for symphonies as compared to other broadcasts, is the fact that the operator or producer has little say-so as to physical arrangement of the orchestra. This arrangement is dependent upon the conductor's whims, and there are few who would allow any change from desired arrangement. The new operator should be tolerant of such an attitude, since the emotional interpretation of a symphony score is utterly dependent upon the conductor and how he hears the individual and blended tones from the great number of instruments. It will be found in practice that little difference is made anyway in physical arrangement of a large orchestra, since acoustical conditions are usually optimum wherever such organizations perform.

The first information to get when preparing a symphony setup is the program for the broadcast. This should include data on whether or not a choral group is combined with the orchestra and types of solos, (piano, vocal, etc.). The next thing to do is to contact the conductor personally (in some cases the symphony business manager must be contacted instead of the conductor), and ascertain if any definite microphone placement is preferred or demanded as dictated by previous experiments or broadcasts. We are then ready to proceed with the preliminary microphone setup, as outlined above.

In case of a piano solo, with intermittent orchestral accompaniment, no change need be made from the location of the main orchestra microphone position since this nearly always proves ideal for the piano. Usually the piano is placed in front of the first violin rows and placed on full stick, which is common practice for good audience listening in such numbers.

Arrangements with a single voice accompaniment almost always require a separate mike for the vocalist. Remember, however, that this voice must be given auditory presence, while maintaining the auditory presence of the orchestral accompaniment.

Remember, too, the listening technique of the symphony listener and the added fact that music is the emphasized interest of the symphony program. Announcers should be peaked lower than zero vu by an amount compatible with these conditions. "Gain riding" is to be held to an absolute minimum; any necessary adjustment of fader control should be carried out in a gradual, undiscernible manner.

Organ Pickups

Pipe organs are seldom used in studios today, having been largely replaced by the electronic type. In a mike setup for pipe organ, sufficient distance must be maintained to get a good blend of tones, and will naturally be governed by the particular acoustics involved. "Cut and try" is about all that can be honestly written about this.

Electronic organs are very commonly used in studios. There are two general types of reproducer loudspeakers associated; the older type with openings on both sides and at top (the major opening is

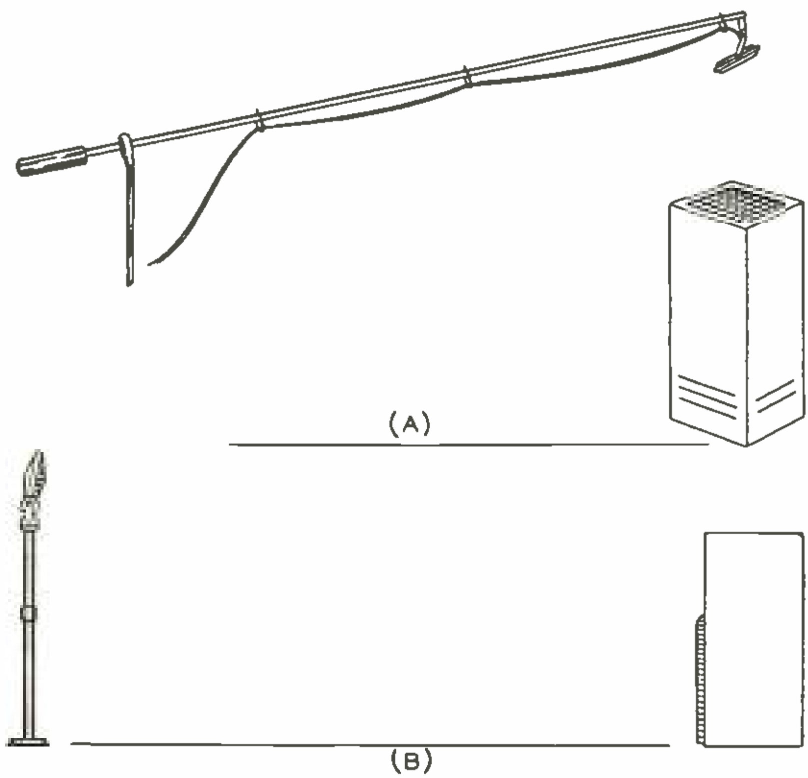


Fig. 4-17. Microphone pickup for electronic organs with the major loudspeaker at the top of the organ (A), and for the loudspeaker opening in the front (B).

on top) and the newer type that resembles closely the ordinary loudspeaker enclosure with opening only at the front of the cabinet.

Fig. 4-17 illustrates the best arrangement for those two distinct types of reproducers and is self-explanatory.

Choral Pickups

The wide dynamic range as well as the size of large choral groups emphasize the importance of getting the mike back as far as the acoustical conditions permit in most ordinary studios. In very large studios, the mike should be back far enough that individual voices or sections are not emphasized. It is absolutely essential that risers be used for any large vocal group. When rows of singer's are all on the same level, even a microphone raised high by means of a boom stand will not give satisfactory results.

See Fig. 9-3 (Chapter 9) for proper sectional arrangement of vocal groups for broadcasting. The reasons for this arrangement are discussed in the accompanying caption of Figs. 9-2 and 9-3 of Chapter 9.

Importance of Rehearsals

The co-ordination of hand, ear, sight, and sound for the purpose of blending the component parts of a studio performance is best gained by the operator through the medium of rehearsals. Fading in or out of various microphones, turning them off and on, is the procedure with enables the engineer to play upon the sound of voice and orchestra much as if the control panel itself were a musical instrument. Indeed, in a sense, this is just what it is. The ratio of fader adjustments will determine the apparent distance of a singer from the audience; the voice may be smothered with music or may be made to stand alone with only a suggestion of background accompaniment. A proper blend of voice and music, or of dramatics and sound effects, can only be properly created through careful and detailed rehearsals. This is the one and only method of preventing the "on air" show from becoming only a caricature of the original idea.

Many "old timers" are familiar with the coloratura soprano who is nicely "adjusted" on rehearsal, then hits +20 vu on the air without batting an eyelash. This condition simply emphasizes one important point: the operator must be apt at diplomacy as well as technically conscious. Talent *must* be made cognizant of the importance of treating rehearsals just the same as "on air" performances. If the performers are instructed in "mike technique" from the point of view of making *their* performance sound just the way *they* desire to the listener, the operator will find ready and willing co-operation. Do not be shy of temperament. The more temperamental the performer, the more he likes to be "fussed over" at rehearsals to gain emphasis of his best talents. Ask any operator or producer of big time shows out of New York, Hollywood, or Chicago; they are in a position to know.

The distance to be maintained between vocalist and microphone will depend on the type or style of vocal form used by the singer. In general there are two commonly encountered types of vocalists, the "crooner" and the "operatic" singer. Whereas the crooner will employ a dynamic range of around 15 vu, the "operatic" singer will use a much wider dynamic range. For the former type, where the sound waves are garnered principally from activation of the upper larynx and throat muscles with comparatively low-pressure waves resulting, it is usually necessary to work close to very close into the microphone. The vocalist who "sings out" by bringing the chest muscles into action must be placed a minimum of 4 feet, preferably 6 to 8 feet, from the microphone. This may appear to be an excessive distance, but actually a much greater dynamic range and brilliancy of voice may be realized by using this distance for singers who range from extremely low to very high air pressures to excite the microphone element. There are, of course, a number of "in between" singers, such as some of those who sing with dance orchestras, and they are usually placed from 2 to 3 feet from the microphone.

Microphone technique for actors in a dramatic program spells success or failure in creating the desired illusion in the loudspeaker. Usually one microphone only is used for the entire cast, with a separate microphone for sound effects. As each actor plays his part, he steps up to the microphone, sometimes approaching from the fading zone into the announce zone to create the illusion of approaching the scene of action, sometimes leaving in the same manner. In some cases "board fades" are marked on the script (Fig. 4-18). The operator fades the entire studio setup including sound effects by fading out with the "master gain" control. Shouts or screams must be performed in the shading area "off mike" to avoid excessive pressure on the microphone element which would require an excessive gain adjustment by the operator, losing the effectiveness of the illusion.

While studio rehearsals are in progress, it is imperative that the engineer and production director be able to talk to the cast for the purpose of instruction in positions, microphone technique, etc. This is accomplished by means of a "talk-back," which consists of a microphone in the control room connected to an amplifier feeding a loudspeaker in the studio. Switches on the control console and perhaps also on the production director's console where such is used, are pro-

1. OPENING SOUND. SOMBER, DRAMATIC STRAINS (ORCHESTRA)
2. NARRATOR. (LOW MONOTONE, VERY CLOSE TO MIKE). Have you ever been
3. to Hell?... Well I have ... and now I have to go back...
4. to stay!
5. SOUND. MUSICAL CRESCENDO - THEN SILENCE FOR 3 SECONDS...
6. NARRATOR. She had everything a man could want, topped off with
7. a beautiful name ... Clarissa. I'll never forget
8. that first night I saw her, it was raining... as it
9. had been for hours... and..
10. SOUND. FADE IN RAIN ON PAVEMENT. STREET NOISES..AUTO HORNS.
11. CAR SWIFTLY PASSING..SPLASHING WATER..
12. CLARISSA. (STARTLED) (OFF MIKE) Oh!
13. NARRATOR. Oh Miss! That's a tough break... Better get back here..
14. farther from the curb.. motorists don't think you know.
15. Here..share my umbrella.
16. CLARISSA. (DRY LAUGH). I'm afraid I shan't get any wetter now..
17. Thank you anyway. (PAUSE) My name's Clarissa, what's
18. yours? (START BOARD FADE HERE) 3 SECOND PAUSE.
19. NARRATOR. (REMINISCENTLY). Just like that... and she was young..
20. and so sweet... and beautiful!

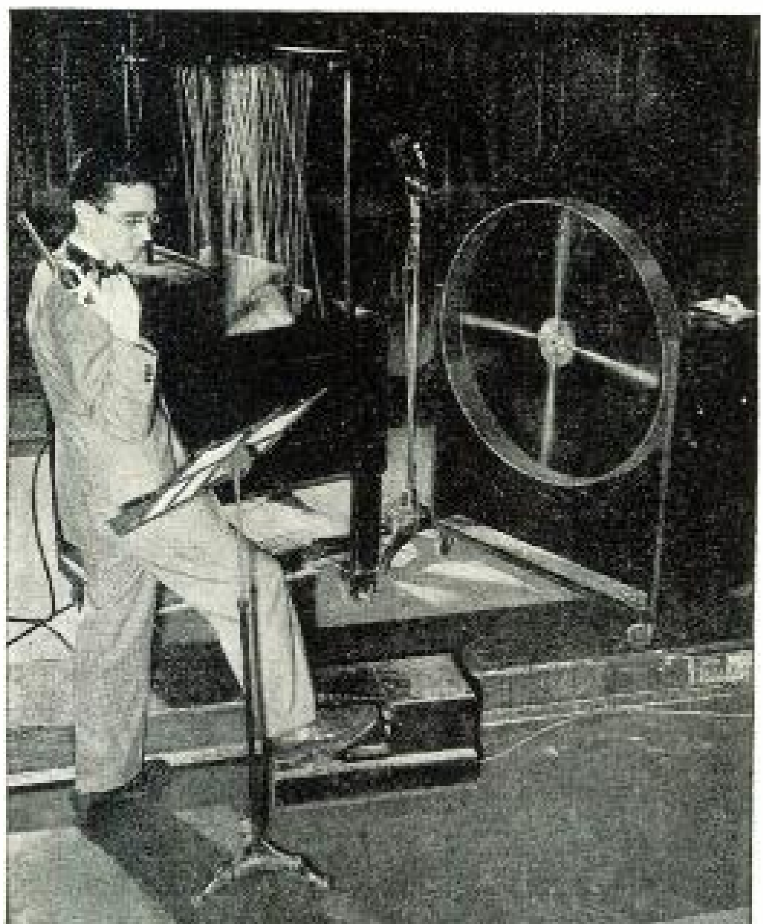
Board Fade

Fig. 4-18. Sample of script which an engineer has marked so that he can "cue" himself for what is coming. The "board fade" is done by fading the master gain control. Note that in line 10 the "fade in" might be done by the turntable operator if sound is on records; if rain and street sounds originate in studio, control man fades in associated microphone.

vided for the talk-back mike. The control man is sometimes provided with a foot-switch to free his hands for the controls. When this mike is turned on, the control-room speaker is cut off by a relay, interlocked with the switch, to prevent acoustic feedback. This microphone is also electrically interlocked with the "on-air" position of the output switch so that it may not be operated during the time a show is actually being broadcast.

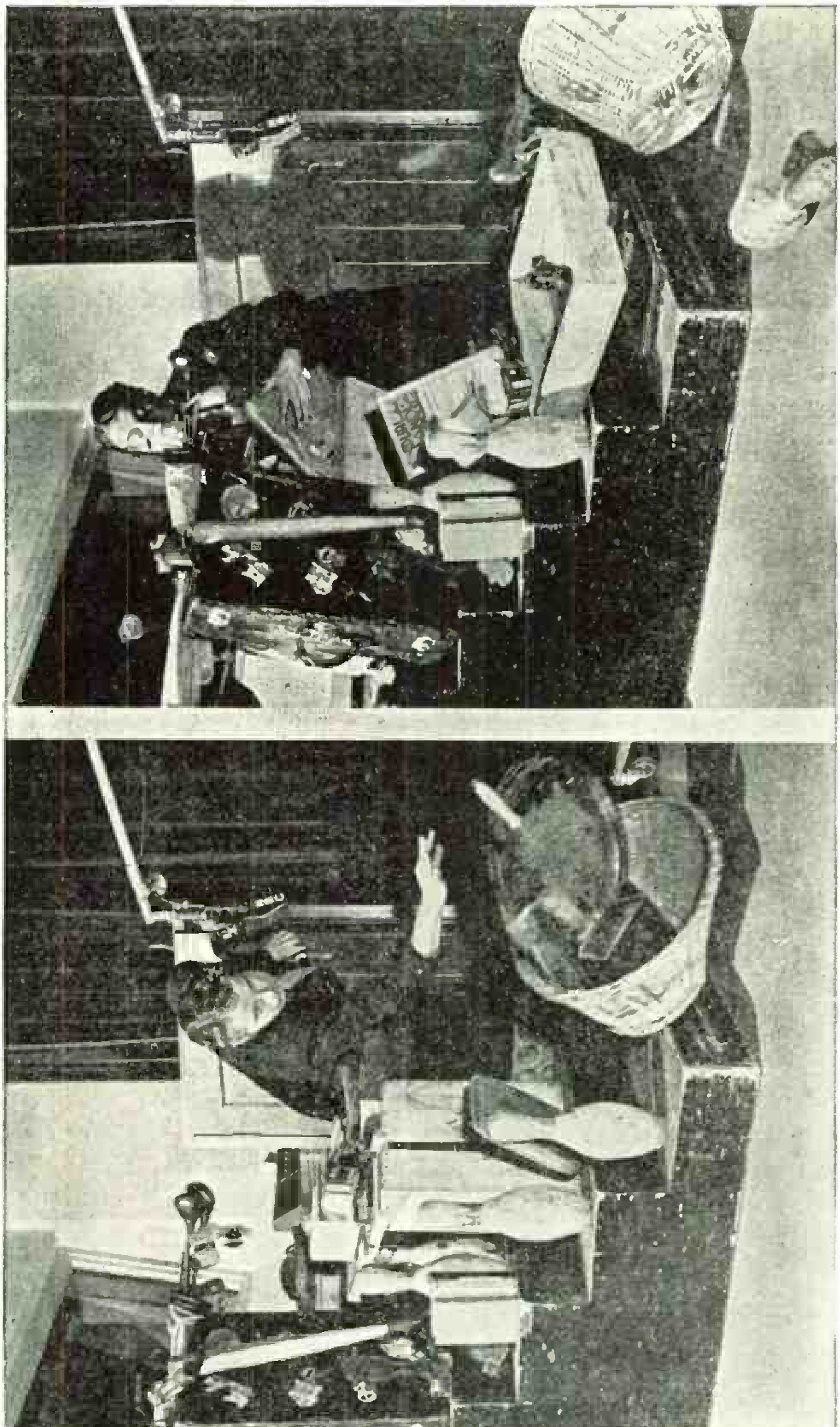
Sound Effects

Major network practices in the art of sound effects has developed over the years into one of the most highly specialized fields of broadcasting. A sound effects technician can take the lowly strawberry box and create illusions ranging from the squeak of a wooden gate or the squeak of a ship moored to a dock, to the terrible rending crashes and splintering of wood for collisions of any description. A bow of a bass viol is drawn in a particular manner over the edges of the box for the first effects, while the box is crumbled between the hands close to the microphone for the collision effect. Rainfall is simulated by the pouring of birdseed or buckshot on a sheet of parchment or by a rain ma-



Courtesy NBC

Fig. 4-19. Sound effects for a good night for a murder. The noise of the howling wind comes from the electrically revolved reeds in the circular shield at the right, controlled by the operator's foot, and the heavy rainstorm results from the rain machine on the left, being turned on full force.



Courtesy NBC

Fig. 4-20(A), left, (B). When the door of the famous closet of the "Fibber McGee and Molly" program is opened by the operator (A), he starts the noise by pushing the clothes basket and its contents off the lower step. In (B) he is shown just before shoving the rest of the din-making items off the upper steps, the noise being picked up by the microphone suspended at the right.

chine, consisting of perforated pipes through which water pours onto brushes in a tub, as shown in Fig. 4-19. Of course, actual objects are also used to produce certain sound effects as illustrated in Figs. 4-20(A) and (B), which show the contents falling out of Fibber McGee's famous closet on the NBC "Fibber McGee and Molly" program. BB shot rolled back and forth with skillful timing over a copper screen can simulate either a lazy palm-bordered beach or a veritable turmoil of angry waves in an ocean storm. Cellophane crackled gently between the hands close to the microphone can create the illusion of the most terrible forest fire imaginable.

The sound technicians' heterogeneous collection as shown in Fig. 4-21, consists of all sorts of weird machines, hail and wind machines, boxes in which glass is shattered, thunder drums, hurricane machines, heavy doors on frames, keys, and a thousand items entirely beyond the scope of this book to reveal. In addition he has a console on which a number of turntables are mounted with their individual pickup arms and dials which automatically "count" the number of grooves set in from the edge for proper "cuing" sound effects. These turntables may

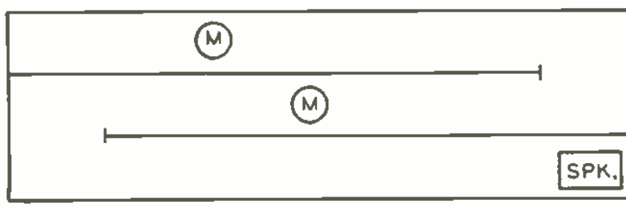
Courtesy NBC

Fig. 4-21. The sound equipment storage room in the Chicago studios of the National Broadcasting Co.

be varied from 0 to about 150 rpm to make still more flexible the number of weird and uncanny effects that can be obtained from recordings.

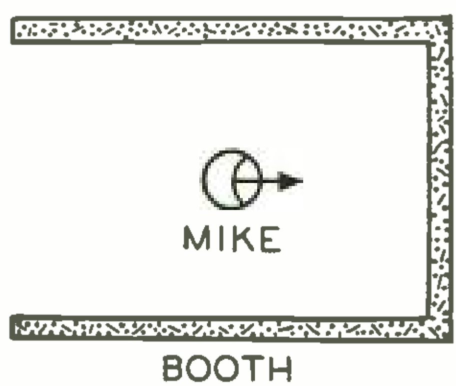
The voice can be made to sound as though it were coming over a telephone by means of a "filter mike," which is simply a microphone run through a filter amplifier, clipping high and low frequencies so that the quality is similar to that heard in the telephone receiver. Reverberation may be added by feeding the signal to be so treated into a speaker at one end of a "reverberation chamber" as illustrated in Fig. 4-22. This type of reverberation setup is being largely replaced by a continuous tape arrangement by which any amount of reverbera-

Fig. 4-22. The further a microphone is placed from the speaker in a "reverberation chamber," the larger seems the hall or cavern in which the action occurs.



tion may be added to the original sound synthetically. The farther away the microphone is placed, the larger becomes the hall or cavern meant to be simulated in the drama. The illusion of talking in close quarters such as that of a telephone booth is created by placing a microphone in a sound absorbent booth, as shown in Fig 4-23.

Fig. 4-23. When a microphone is placed in a booth with walls lined with sound-absorbent material, the resulting speech seems to be coming from a telephone booth.



Although the average broadcast station is much more limited in elaborate equipment utilized by the major network key stations, there is no limit to the possibilities of using what equipment is available to the ingenious technician. A good telephone effect can be achieved by feeding a microphone signal through a separate amplifier and exciting a pair of cheap headphones (the cheaper, the better) that may be held immediately adjacent to another microphone in the studio. This circuit is illustrated schematically in Fig. 4-24.

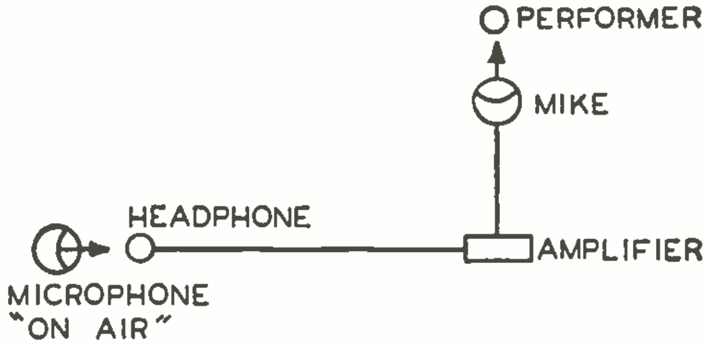


Fig. 4-24. Conversation over a telephone can also be simulated by feeding the amplified output of one microphone into headphones and broadcasting their sound.

A slight reverberation effect may be obtained by placing a microphone immediately above the sounding board holes of a piano, and directing the voice by means of a megaphone or tube over the strings of the piano. The sustaining pedal is held down to allow the strings to vibrate freely when the voice waves are impinged upon them.

Several good textbooks exist on sound effects and will provide more detailed information.

Special Effects

Special effects, as distinguished from sound effects, are often called for in modern broadcast programming. Many such effects are achieved by special techniques in disk and tape recording, requiring complex "dubbing" procedures to unify the program.

Perhaps the most popular example to illustrate this type of effect is the elfin voices and music heard on certain children's recordings. To get the elfin voice effect, the orchestral background was recorded first on 78 rpm disk. This was then played back on 33⅓ speed, and the voice of the singers was blended to this tempo on another 33⅓ recording. When this recording is played back at 78 rpm, the musical background returns to normal sound, while the voices are obviously stepped up in tempo and pitch, resulting in the elfin-like voices so loved by children.

There are many variations of special recording techniques for special effects. Recording of musical backgrounds and vocal accompaniment on separate tapes allow adding reverberation to either vocal or background at points in the score where this is effective. Reverberation may be added to either one or both simultaneously, or varied at will. Many late recordings have made use of such special effects.

Transcription Turntables

The practice of playing recordings and transcriptions varies considerably with different stations. In many of the smaller stations, the control man operates the turntables as well as running the control console. In the majority of the stations of 5 kw or more, either the announcer runs the tables, or an especially trained person is used to run the turntables, which may be in a separate room just for this purpose, as shown in Fig. 4-25.

Recorded and transcribed shows constitute a most important part of a broadcaster's daily schedule. "Transcriptions" are recordings made especially for broadcasting purposes; they are usually 16 inches in diameter and use a turntable speed of 33⅓ rpm to enable recording a full 15 minutes of program time.

Courtesy WOR

Fig. 4-25. The transcription studio at Station WOR, New York.

A transcription "platter" may also consist of a number of separate musical or voice selections on a single disk, in which case they are numbered on the label with the titles of each number listed. Also on this label will be the information as to lateral or vertical cut, start on inside or outside groove, and reproduction speed (33⅓ or 78 rpm). This is enough to keep any operator "on his toes," especially when a program consists of both recordings and transcriptions which may require change of turntable speed, lateral or vertical switch placement, and noting whether the cut is started on the inside or outside groove.

Then too, a filter selector switch is employed to select a suitable frequency compensation for the particular disk used. For example, RCA lists typical switch positions for the 70-CI turntable as follows:

Lateral

#1. Transcriptions, Orthacoustic, Columbia
#2. Home records and worn transcriptions
#3. Home records, World, Decca, and AMP
#4. Test records and special recordings (wide open at highs).

Vertical

#1. World and AMP transcriptions
#2. Worn transcriptions.

All records and some transcriptions, which are played at 78 rpm (same as the record player at home), are "laterally" cut, and played from outside groove toward the inside. Most transcriptions, however, play at 33⅓ rpm. In addition to this, some transcriptions are cut "vertically," that is, the groove variations that comprise the signal are varied up and down instead of side to side, using the depth on the coating of the disk. Also, some of them play from inside-out, and require the starting of the pickup arm on the inside groove.

Turntable Operation

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 (lateral or vertical); that the turntable speed switch is on the correct speed adjustment for the particular recording used (33⅓ or 78 rpm), and that the disk has been properly "cued." This means that the pickup arm must be at the spot on the groove where the announcement or music begins, so that no time is lost in waiting for the arm to reach that point on the disk. This is usually accomplished by using headphones on an auxiliary amplifier so that each disk may be "cued in" preparatory to going on the air.

When the disk has been properly "cued in," most experienced operators find it advantageous to start the turntable moving while holding the disk (on the outside edge so as not to touch the grooves) to keep it from turning until start is desired. This practice eliminates "wows" that are apt to occur on the starting due to time taken for the turntable to gain proper running speed. When this trick of operation is not followed, the disk should be "cued back" at least one full revolution of the turntable so that the proper speed will be reached before start of the signal.

The art of smooth turntable operation on the air takes considerable practice by the operator. Familiarity with the operating procedure can be gained only by practice, and most stations demand a thorough "break in" training before entrusting the operator with an air show comprised of recordings and transcriptions.

The music library of a broadcast station may contain files of thousands of recordings and transcriptions, and their proper care in storing and handling is an important factor in "on the air" quality of reproduction. Excessive heat and dust in the air are major enemies to be considered in the storage room. The library should be well air-conditioned, with an efficient dust-filtering system. In any case, the disk should be cleaned with a soft dry cloth before playing. Static electricity causes dust to cling tightly to records, and all precautions such as use of linoleum floors in library and turntable room to reduce static electricity should be taken. Finger marks cause noisy reproduction due to the oil and grease from the hands causing foreign matter to cling close to the walls of the grooves. Platers should be handled on the edge only.

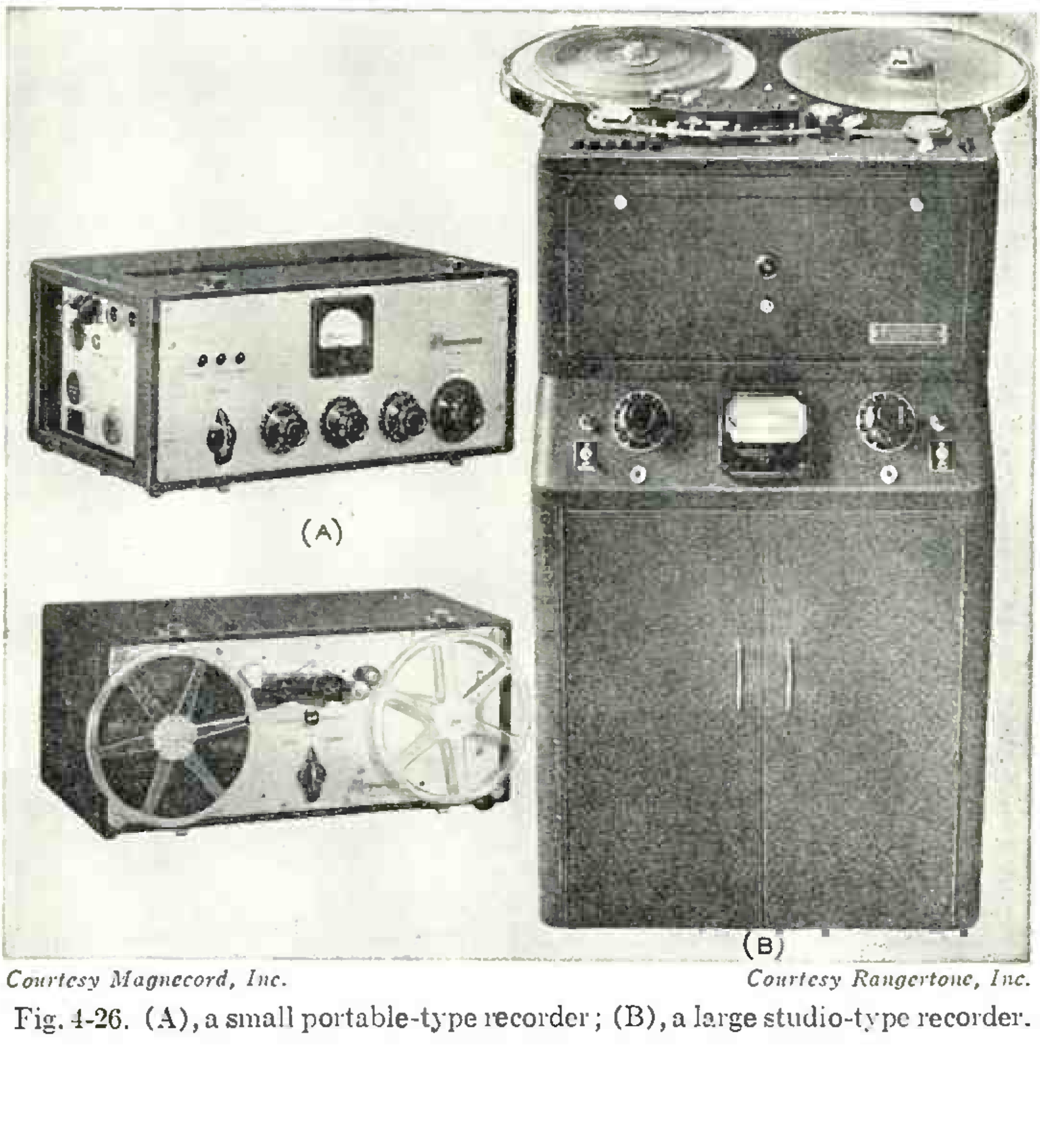
For this same reason, the permanent type pickup needle should not be "swiped-off" with the fingers in an attempt to clear it of dust. A small soft brush should be used.

Magnetic Wire and Tape Recording

The most recent popular application of storing sound energy for later reproduction purposes is the magnetic wire or tape method. This system holds startling possibilities in an almost infinite field of application, and the immediate future will undoubtedly embrace a tremendous expansion of the use of the magnetic method of recording and reproduction.

This means of retaining sound in the form of a magnetic property of wire or tape offers a large number of advantages over disk recording for many types of applications. Once a magnetic recorder has been properly serviced and prepared, there are no delicate adjustments such as stylus angle and depth characteristics. There is, of course, no processing necessary before playback. Wearing out of the material with increase of noise and distortion with use, is infinitely less than that of an instantaneous recording, and much less than that of a commercial record. A much wider frequency range may be employed without special equipment and ultra delicate adjustments and handling. Modern commercial tape recorders easily handle the entire range of 30 to 15,000 cps, sufficient for use in f-m broadcasting. An extremely low noise level is more easily achieved, allowing cleaner reproduction and extended dynamic range. The recordings are easy to handle and store; there are no warpage problems, very little deterioration, no breakage in storage, and are easily mailed without extreme complications in packing. In fact, it is outside the scope of this text to cover fully the engineering and practical advantages of magnetic recording over the disk method. *Disk recordings, of course, will still remain necessary for choice of short, individual musical selections in the home or broadcast fields.*

Two representative types of magnetic tape recorders are shown in Figs. 4-26(A) and (B).



Courtesy Magnecord, Inc. *Courtesy Rangertone, Inc.*

Fig. 4-26. (A), a small portable-type recorder; (B), a large studio-type recorder.

How Magnetism Is Used to "Store" Sound

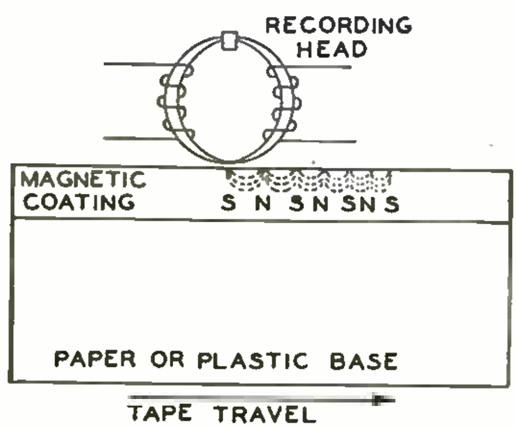
Here at last is a method of recording and reproduction of sound that *does not* require a conversion or reconversion of sound energy into corresponding physical movement of a mechanical mass. Microphones, disk recorders, and playback all utilize a moving physical mass, that must (ideally) respond in the same way over the usable frequency range of the system.

The principle of magnetic recording may be simply defined as follows:

Electrical energy is converted into magnetic force. This magnetic force is impressed upon a moving magnetic wire or tape, which may then be utilized to reconvert the recorded magnetic energy back to electrical energy.

So that the process of this method will be better understood, it is well to review briefly some modern theory of magnetism. A magnetic field may be created by an electric current flow; that is, a magnetic field exists around the conductor of an electric current. When a magnetizable substance is brought within this field, the small atomic regions or "domains" of the material *will have their axis of position shifted in alignment with the field of force about the electric current, and will then exhibit a magnetic field of its own.* When the current flow is stopped, or the magnetizable substance is withdrawn from the field, it will retain this property of a magnetic force of its own to an extent depending upon the *remanence* of the material.

Fig. 4-27. The magnetizing action of a magnetic recording head on the magnetically-coated paper or plastic tape used in magnetic recording is shown here. The magnetizing gap in the electromagnetic head of the recorder, shown close to the surface of the tape, lines up the small metallic particles in the coating of the tape, producing an infinitely large number of small magnets of varying degrees of magnetic strength depending upon the varying strength of the recording signal.



Although wire recorders use a very thin steel wire as the recording medium, magnetic *tape* methods no longer use actual steel as the substance. A paper or plastic base is used, upon which is placed a magnetic coating. This tape is ¼ inch wide, and usually about two thousandths of an inch thick. The actual magnetic coating is about one-half thousandths of an inch deep.

Fig. 4-27 illustrates the most basic action of a magnetic tape system. As the signal is fed into the recorder head, the tape passes at a constant speed across the very small air gap of the recorder head. This air gap may be as tiny as 0.0005 inch. (Influence of gap size is mentioned later.) Thus at any given instant, a tiny magnet of this dimension is formed, and the recorded signal consists of a string of these minute longitudinal magnets whose flux and polarity correspond to the signal current in the head at the instant the magnet was formed. The action of wire recording is the same in principle as the above.

The magnetic properties of a signal recorded on magnetic tape may be illustrated as shown in Fig. 4-28. The sound track here has been made visible by means of a method similar to the mapping of fields about a bar magnet by using iron filings; a phenomena familiar to most readers. In this instance, the recorded tape was passed through a suspension of carbonyl iron in heptane. Carbonyl iron is particles of iron much smaller than ordinary iron filings. It may be observed by this illustration that the tape actually consists of a string of magnetic fields which correspond to the signal recorded.

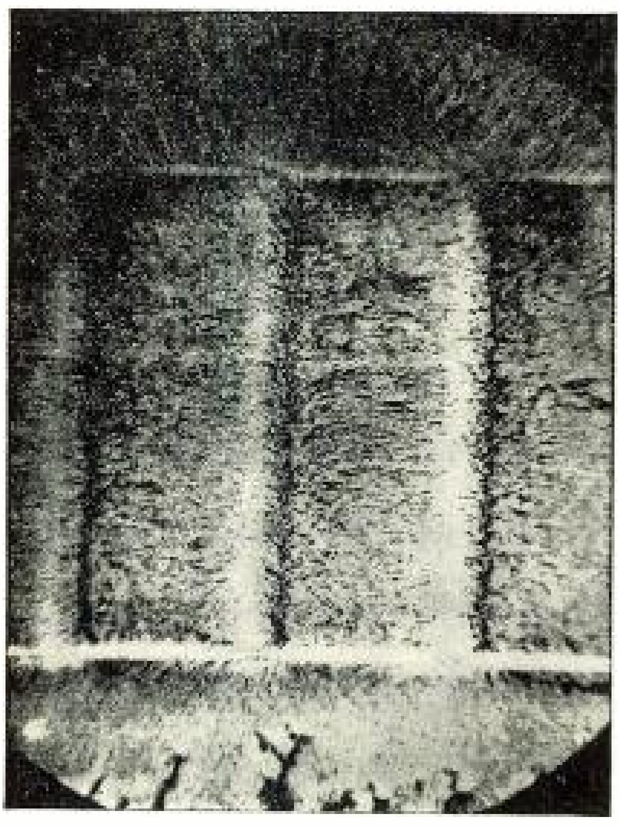


Fig. 4-28. Sound track of a signal on a magnetic tape, made visible through the use of carbonyl iron (iron particles). Shown in this track is a low-frequency note recorded on 120 mil track width. The magnification is 20 times.

Courtesy Minnesota Mining and Mfg. Co.

Tape recorders have a large number of advantages over wire, among them being the following factors:

1. Cannot twist or tangle (better fidelity and ease of handling.)
2. Splices with Scotch tape, simplifying editing or repairing accidentally broken tape.
3. Splices are not detectable, whereas wire splicing is audible.
4. Lower crosstalk in storing. (Wire, since it is not insulated such as tape with its paper or plastic base, imparts its magnetic properties to adjacent layers.)
5. Lower speed for a given frequency range than wire.
6. Complete electronic erasure more easily attained.

Basic Principles of the Tape Recorder

Fig. 4-29 show the most usual arrangement of a tape recorder. This is known as the "capstan drive," used in all tape recorders where reliable and flutter-free operation is mandatory. The capstan is driven by the drive motor, and is weighted with a flywheel to assure excellent regulation. Tension on the tape is such that it is held firmly against the capstan. The supply reel, containing the tape is at the left, and is also actuated by the rewind motor to rewind the tape after recording. The tape is threaded through the erase, record, and playback heads, around the capstan and into the takeup reel. This reel is also driven by the forward motor. The speed of rotation may be adjusted either by variable speed motors or by varying the size of the capstan.

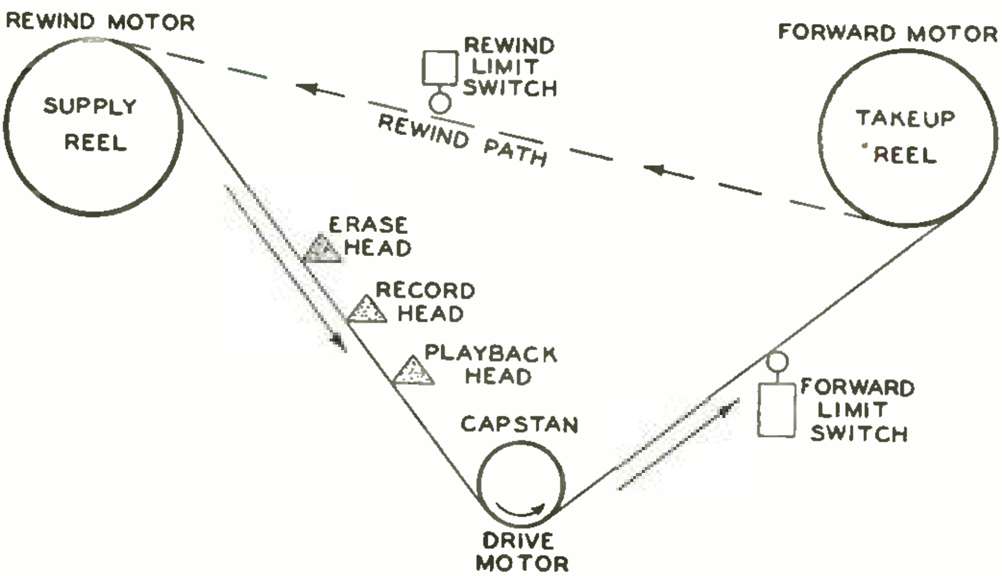


Fig. 4-29. A typical capistan-drive type magnetic tape recorder.

A little thought will show the importance of tape speed to the matter of fidelity. The wavelength of the recorded signal for a given frequency will depend upon tape speed. For a 15,000-cps tone on a tape moving at 15 inches per second, one full wavelength of this recorded signal will be only 0.001 inch long. Now assume that the air gap of the magnetic head was just 0.001 inch. This would constitute a span of one wavelength of the signal and such a tone would not be reproduced. Thus for this tape speed, the gap should be no more than one-half wavelength of the highest frequency to be recorded, or 0.0005 inch. For a gap of 0.001 inch, it would be neccssary to run the tape at 30 inches per second to adequately cover a frequency range up to 15,000 cps. Thus it should now be possible for the reader to picture the relationship between tape speed, head gap size, and required fidelity. Extremely small airgaps require micro-inch precision and expensive machining.

Although no standards have definitely been established for magnetic recording at the time of this writing, there are trends to standardization in the industry. The two most popular speeds of tape recording are 7.5 and 15 inches per second. Modern high-quality commercial tape recorders can reproduce 50 to 15,000 cps at a tape speed of 15 inches per second. Up to 7,000 cps may be covered at a speed of 7.5 inches per second. A reel of tape that will accommodate 33 minutes of playing time at 15 inches per second speed, will accommodate 66 minutes of playing time at 7.5 inches per second.

Wire recorders use a speed of 24 inches per second, and rarely attain a better range than 100 to 8,000 cps.

The Necessity for Tape Bias

It should be remembered from magnetic theory that the extent of magnetization will depend upon the strength of the magnetic field. Since this is so, the amount of magnetism retained will depend upon the magnitude of the original magnetic field for a substance of given remanence, or ability to retain magnetic force.

The curve of Fig. 4-30 shows a plot of the amount of remanence against the strength of the signal field. This curve may be compared in effect to the plate current of a vacuum tube as plotted against grid voltage. It should be obvious here to the reader familiar with amplifier theory that there is no suitable straight portion of the curve that would result in an undistorted signal output. Fig. 4-31 shows the

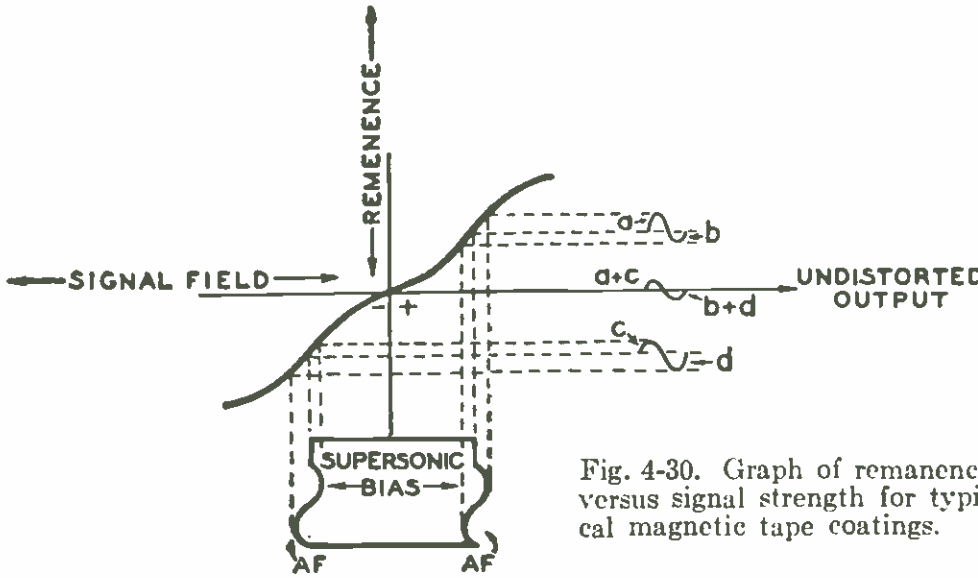
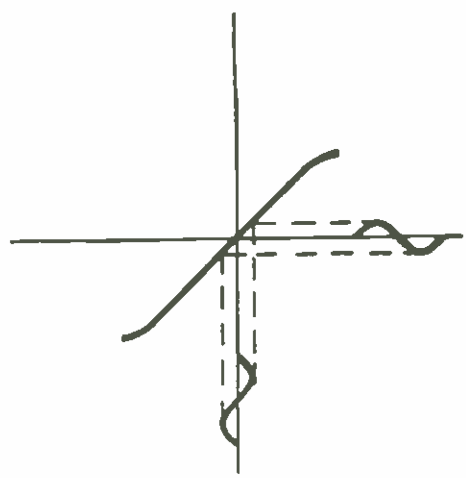


Fig. 4-30. Graph of remanence versus signal strength for typical magnetic tape coatings.

type of curve necessary for undistorted output. Unfortunately, the curve of Fig. 4-30 is the type of curve encountered in magnetic materials. The problem of obtaining an undistorted signal voltage from such an operational characteristic is the matter under discussion at this point.

Fig. 4-31. If the remanence curve for the magnetic coatings of magnetic tapes were linear as shown here, linearity of reproduction would be obtained as is shown by the undistorted sine-wave reproduction of the original sine wave.



Since there is no practical straight portion of such a magnetization curve, d-c bias is out of the question when using a tape of zero magnetization as the operating axis. To overcome this difficulty, a high-frequency bias is used. Since this bias current is far above the highest audible frequency to be used, it is given the term "supersonic bias." This bias voltage is *not modulated* by the signal current, but rather is mixed together to form a pattern shown in Fig. 4-30. Fig. 4-32 shows the difference of a modulated pattern with that of a supersonic and a-f pattern as used in tape recorders.

Study now the information shown in Fig. 4-30. The audio component on the "plus" side of the zero magnetization axis results in the distorted output waveform shown in the upper portion, when the *a* loop is larger than the *b* loop. The audio component on the "minus" side results in the lower distorted waveform where the *c* loop is smaller than the *d* loop. Since these complete loops are combined on the axis of operation, the resulting output signal is undistorted as shown in the figure. That is, the algebraic sum of *a* plus *c* will equal the algebraic sum of *b* plus *d* when the bias is of the proper magnitude for the particular installation and type of magnetic tape used.

Fig. 4-32. Difference between a modulated waveshape and a *combination* of the supersonic and audio waveshapes as used in magnetic recording.

While the amount of bias current is critical, as will be shown shortly, the actual frequency of the bias does not seem to be very critical, as long as the frequency is high enough not to beat with the harmonics of the audio current. In practice, it is customary to make the bias frequency five times the highest audio frequency to be reproduced plus 5 kc. Thus a tape recorder of a frequency range up to 5,000 cps would use a bias frequency of five times 5 or 25 kc, plus 5, equaling 30 kc. A recorder system utilizing the full range up to 15,000 cps would, therefore, require a bias frequency of at least 80 kc. Commercial makes of tape recorders utilize a bias ranging from 30 to 100 kc. Higher frequencies of bias seem to give better signal-to-noise ratio on the average than lower bias frequencies, and most manufacturers are now using the higher frequency values even for home-type recorders.

Effect of Bias Magnitude

Bias magnitude, or amount of bias used (measured by bias current readings or ampere turns of bias) is a very important and critical characteristic of magnetic recording. Too little bias results in high noise and distortion levels. Too much bias results in loss of high frequencies.

In practice, however, bias is adjusted for minimum noise and distortion; any resultant loss of highs is compensated for by increasing tape speeds or equalization procedures.

The Minnesota Mining and Manufacturing Company,[1] one of the leading researchers in magnetic-tape characteristics, have conducted exhaustive tests on the effect of bias on their particular sound tapes #100 and #111. The first is a paper base magnetic tape, the latter using a plastic base. As a basis from which to start, they first ran tests on the effect of bias current on the output they could obtain with no more than 1% third-harmonic distortion. Measurement of the third-harmonic distortion was chosen in preference to total distortion, since the total measurement would be affected by noise level. Measuring the third-harmonic distortion theoretically relates the measurement only to the magnetic properties of the medium.[2] *Even* harmonic distortion has been found to be negligible when using magnetically neutral tape (a-c neutralized tape instead of d-c biased or saturated tape), with a supersonic bias of good waveform. This will be discussed more in detail presently.

The curves of Figs. 4-33(A) and 4-33(B) show the results of the manufacturers' tests on effect of bias with the "Scotch" sound tapes #100 and #111, respectively. For the #100 tape (paper base), it may be seen that the highest output obtainable with no more than 1% third-harmonic distortion results from a bias value of 3 ampere turns (solid line). Actual number of bias ampere turns is dependent upon the particular recording head used (the Brush head is used in this case), but the effect of changes of bias are substantially the same with any head. (Ampere turns are the product of the current through the coil in amperes, times the number of turns.)

Fig. 4-33(B) shows that the optimum bias for this test on the #111 sound tape (plastic base) results from 2.4 ampere turns.

Now observe Figs. 4-33(C) and (D) where the effects of bias on frequency response are shown. It is seen that raising the bias results in severe loss of high-frequency response (probably due to a partial erasing action of stray, bias field). The recording, of course, is unequalized.

It may also be observed from the above studies that using the optimum value of bias to obtain maximum output at no more than 1% third-harmonic distortion, results also in the best high-frequency response relative to low frequencies. It is also known, however, that

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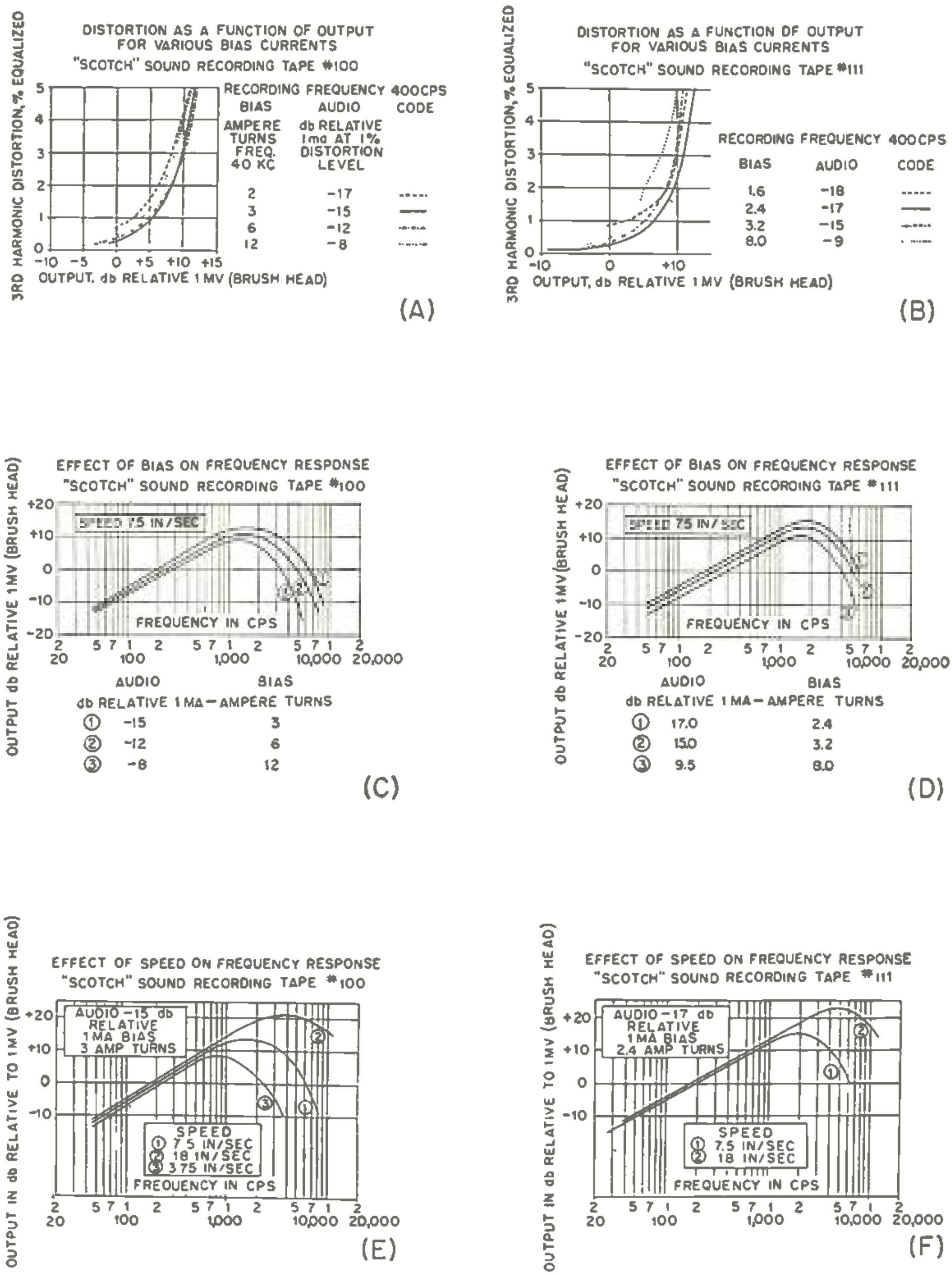
[1] Makers of "Scotch" brand sound tape.

[2] Herr, R., Murphy, B. F., and Wetzel, W. W., "Some distinctive properties of magnetic recording media," *Jour. Soc. Mot. Pic. Eng.*, vol. 52, pp. 77-89, January, 1949.



Courtesy Minnesota Mining and Mfg. Co.

Fig. 4-33. The curves shown here give the relation between distortion and recording-head output (A) and (B), between frequency response and audio bias (C) and (D), and between frequency response and tape speed (E) and (F), for two different types of magnetic recording tapes.

noise level may be *decreased* by using these higher values of bias, resulting in a somewhat better signal-to-noise ratio. The final design must be a compromise for the particular purpose of the recorder, and, as mentioned before, high frequencies may be brought up by equalization, very small air gaps in recording head, or increasing speed of the tape movement.

Effect of Tape Speed

The effect of speed on the recorded wavelength and consequent high-frequency response has been mentioned. The influence of air gap size of the recording head was also shown, and these factors should be kept clearly in mind for a comprehensive picture of magnetic recording.

Figs. 4-33(E) and (F) show the effect of speed on the two types of Scotch sound tape when using the optimum value of bias in the above tests. As borne out by our previous discussion, it may be observed that the highest speed of 18 inches per second results in best high-frequency response. The "high speed" has now been standardized at 15 inches per second. These curves were taken using the Brush recording head.

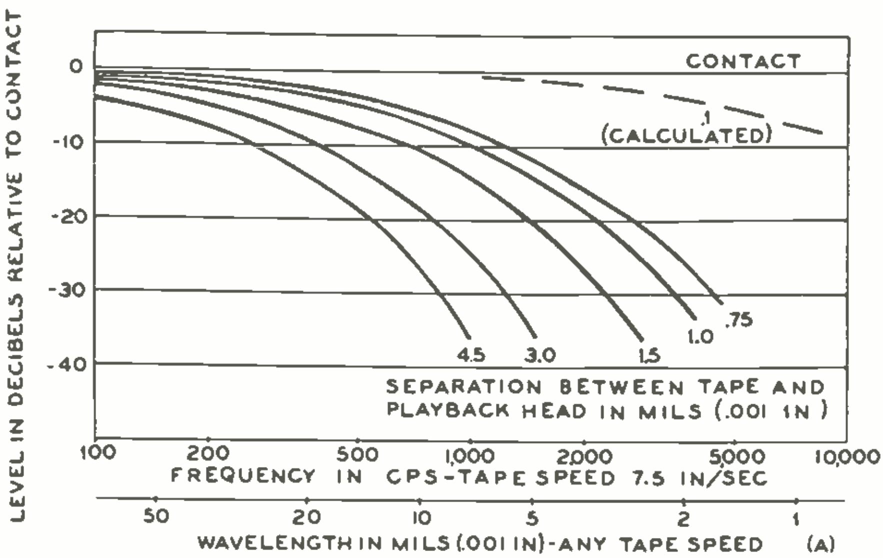
Frequency response is dependent upon quite a few factors other than those mentioned. They may be enumerated briefly as follows:

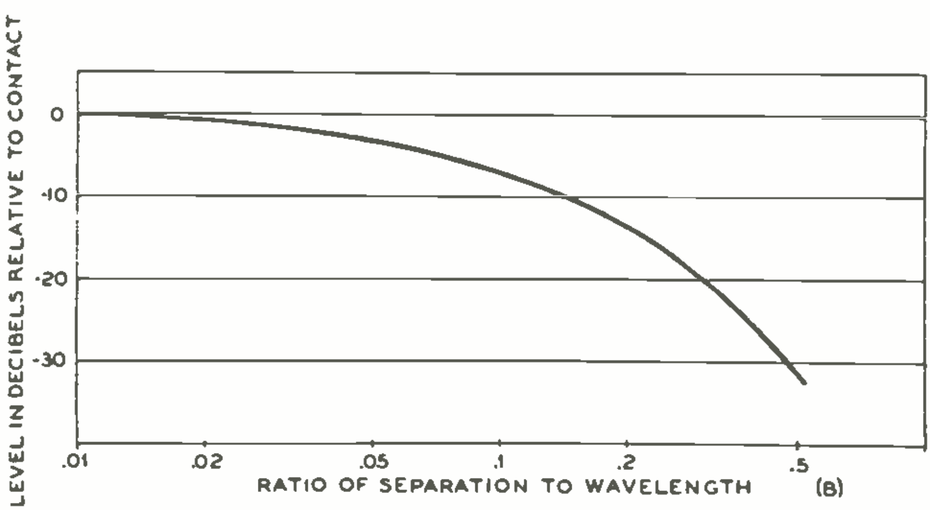
1. The make and design of recording head
2. Recording gap size and design
3. Reproducing head
4. Tape characteristics
5. Azimuthal orientation of heads
6. Characteristics of associated amplifiers
7. Tape contact against pole pieces.

Importance of Tape Contact

In magnetic-tape recorders, it is very important that tape be firmly held against the smooth surface of the pole pieces and across the air gap. We are again indebted to the research work of the makers of Scotch magnetic tape for data showing this importance.

Poor contact between tape and playback pole pieces becomes increasingly harmful at the higher frequencies. Fig. 4-34(A) shows the attenuation caused by various separations of tape and playback head. In these tests, a tape was recorded with various pure tones and then reproduced on a good system with proper contact. Then this tape was reproduced several times with different size paper shims to separate tape and pole pieces by a known amount. The level was then recorded for each frequency and separation, and compared with the original levels. The separation values are given in mils. Data on the frequency scale are plotted as for a tape speed of 7.5 inches per second, but it will be remembered from our previous discussion that the wavelength effect is the important characteristic. In other words, for a given separation, the same attenuation will result at 5,000 cps (7.5 in/sec); at 10,000 cps (15 in/sec); at 2,500 cps (3.75 in/sec); etc. For





Courtesy Minnesota Mining and Mfg. Co.

Fig. 4-34. Two curves showing the attenuation effect of the distance between the tape and playback head for various frequencies and tape speeds.

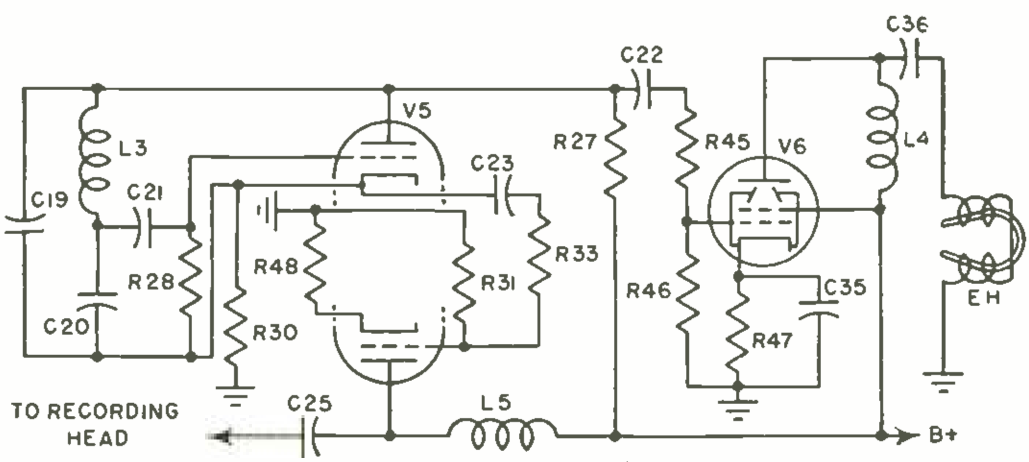
this reason, the wavelength scale is shown below the frequency scale, and this is more significant, being true for any tape speed.

It is observed, then, that attenuation fundamentally depends upon ratio of the separation to the wavelength. The same attenuation results from a 10-mil wavelength and 1-mil separation as for a 45-mil wavelength and 4.5-mil separation. Thus Fig. 4-34(B) was plotted, showing the amount of attenuation against this ratio of separation to wavelength.

It is evident from this data that very small separations can cause serious trouble in attaining comparatively good high-frequency response. The pole pieces must be clean, smooth, and well machined. The tape must be smooth and flexible, with adequate tension or pressure. The Scotch plastic-base tape is superior, in meeting the above tape requirements, to a similar paper-base tape. In addition, the noise level is somewhat lower.

The Bias and Erase Circuit

In modern tape recorders, the erase head is energized automatically whenever the selector switches are placed in the "record" position. readying the apparatus for recording sound on the magnetic tape. The erase head is usually supplied from the same oscillator that supplies the bias current to the recording head.



Courtesy Amplifier Corp. of America

Fig. 4-35. Typical erase and bias oscillator-amplifier circuit.

A typical erase and bias oscillator-amplifier circuit is illustrated in Fig. 4-35. The supersonic oscillator *V5* (a 6SN7 tube) feeds the power amplifier *V6* (a 6V6-GT) which is a parallel plate fed to the erasing head designated *EH* in the diagram. It is absolutely necessary to keep direct current from both erase and record heads to prevent them from becoming magnetized, which causes large even harmonic distortion. (Discussed later.) The (lower) triode section of *V5* serves as an isolation amplifier which supplies a portion of the supersonic current to the recording head for biasing purposes.

Even-Order Harmonic Distortion

This trouble, which is sometimes encountered in tape recording, has been mentioned briefly several times before. To look further into this characteristic, we again may turn to data and research of the manufacturers of Scotch sound recording tape for helpful information.[3]

One of the attractive features of magnetic recording is that the media are inherently free of even harmonic distortion; the magnetization characteristic is not necessarily linear but is symmetric for opposite directions of magnetization. However, in practice, playback wave analysis of a sine wave recorded from a distortion-free source will frequently show appreciable second and higher even-order distortion. When this is found, it is indicative of some defect in the recording equipment, and the purpose of this discussion is to point out what these flaws may be and how to determine and correct them.

A possible source of trouble is in the audio amplifiers, both record and playback. Distortion from this cause may be determined by conventional amplifier testing and eliminated by known techniques not peculiar to magnetic recording.

The causes of even harmonic distortion, which are in the recording process itself, all have one thing in common. This is a d-c component or magnetization which prevents the heads from modulating the tape about the point of symmetry, which is the state of demagnetization. The asymmetric influence may be most clearly visualized in its action at the record head, but if the erase head leaves the tape magnetized, the same results may occur. The troubles which lead to asymmetric recording are as follows.

1. *Permanently Magnetized Heads.* Heads may become magnetized by accidental contact with magnets, by mechanical strains in a magnetic field, or, very commonly, by excessive transient currents occurring in the switching of recording, erasing, or biasing currents.

2. *D-C Components in Erase or Bias Currents.* This is an obvious difficulty; some dubious circuit designs have allowed the d.c. of a

[3] "Sound Tips" Bulletins numbers 3 and 4. Courtesy Minnesota Mining and Manufacturing Co.

tube's plate current to flow through the head, but if such undesirable design is avoided, d.c. can only result from a leaky capacitor or other faulty component.

3. *Asymmetric Bias and/or Erase Waveform.* Even if no d-c component is present, a waveform with different positive and negative peak values will cause the same effect. This is caused by overload or inadequate design in the high-frequency supply.

4. *Stray External D-C Fields.* These will produce the same effect as any of the sources of d-c fields located in the head. Ordinarily the earth's magnetic field is negligible, but serious fields can be caused by magnetized steel parts or especially by nearby meters.

In testing for the existence of asymmetric fields, a small permanent magnet which may be held near the heads is a help. If some position of this magnet reduces even harmonic distortion, the polarity of the magnet should be noted and conditions changed. The heads may be demagnetized by gradual removal from a 60-cps field of about 1,000 oersteds. Bad waveform may be tested by reversing the leads to the heads; if such reconnection changes the amount of distortion or the polarity of the magnet held nearby to improve it, then the waveform is not good. Any steel parts should be demagnetized and any nearby meters temporarily moved to see whether there is an effect. Generally more than one influence is at work with their effects sometimes additive and sometimes subtractive, so it is important to separate the various causes; otherwise very wrong conclusions may be drawn.

It is also worth noting that these same factors which cause even harmonic distortion work to increase noise, and their removal will lower the noise as well as lessen the distortion.

Establishing A-C Bias Values in Magnetic Recorders

Different magnetic tapes, in general, operate best at different values of the a-c bias current. However, the optimum bias for use with Scotch sound recording tapes #100 and #111 are the same. Too low a value of bias may result in high distortion and low signal-to-noise ratio. A bias current set at too high a value results in loss of response at high frequencies.

Except in machines designed for professional use, the bias setting is not particularly critical. In machines designed for home use, an approximation of the optimum value of bias may be had through the use of an audio oscillator, a vacuum-tube voltmeter, and some means for measuring output of the playback system.

Connect the VTVM across the bias windings of the recording head where its reading will be proportional to the bias current. Record a 1-kc note at some level well below the overload point of the tape. Observe the output of the playback system as a function of bias current keeping the recording level fixed. For some value of the bias a maximum output will be observed. The following rules will help select the operating bias in terms of the bias current for maximum output:

1. *For Low-Speed Machines Using D-C Erase.* To obtain the best signal-to-noise ratio, it is necessary to compromise by sacrificing high-frequency response. The optimum bias current is approximately 50% greater than the bias for maximum output.

2. *For Low-Speed Machines Using High-Frequency Erase.* Because a good a-c erase system results in a lower tape noise level than d-c (or permanent magnet) erase, a lower value of bias may be used. Adjust the bias for maximum output of the 1,000-cps signal.

3. *For High-Speed Professional Machines.* In general, more precise techniques for bias setting are required for professional machines. A reasonable compromise, however, is to choose a bias current twice that for maximum output of the 1.000-cps signal.

Common Troubles in Magnetic Recording

Even at this early stage of the development of magnetic recording in this country, there is a large number of different makes of commercial wire and tape recorders. Although this fact embraces a wide variety of physical and electrical designs, the major troubles encountered by the operator have characteristics common to all makes. Some of the more predominating sources of trouble will be discussed here for guidance of the user of any commercial tape recorder.

Distortion. This trouble is, naturally, the most common one encountered in the field, and probably embraces a larger number of possible causes than any other characteristic. Study carefully the previous discussion on bias requirements, and causes of even-order harmonic distortion.

There are three general types of distortion as distinguished easily by the average ear, as follows:

1. Extreme lack of bass and unintelligible speech. This symptom will most likely be caused (assuming that the audio amplifier and microphone are normal) by complete loss of supersonic bias to the

recording head. Check all components in the bias oscillator-amplifier circuit. Check also for loose connections, and check bias winding on recording head.

2. Low output, muffled sound. Again assuming that the audio amplifier is normal, this trouble is usually traced to an accumulation of tape coating residue, dirt, dust, or foreign matter on the pole pieces of the recording head. Clean them thoroughly. This should be a regular part of the maintenance schedule on high-quality magnetic recorders.

3. Little or no output, mushy sound. See No. 2 above. This trouble, however, is most likely an indication of a faulty component part in the audio amplifier or switching system. Check in orthodox manner.

Background Noise. This includes random noise (not contributed by audio amplifier), undesired background sounds and superimposing. The usual culprit in this case is nearly always insufficient or no erase. In many cases, erasing is difficult even with sufficient erase current in the obliterating head, if the previous signal has been recorded on the tape at a very high level. In such cases, the tape should be run through the erase head several times before applying any signal to the recording head.

The most common causes of insufficient or no erasing action may be listed as follows.

1. Defective erase head. Check continuity and physical condition. If defective, replace with new head *of the same type.*

2. Defective contact forward limit switch. Check and adjust contact.

3. Overheating of resistor in erase circuit. If any resistor is running hot (check by feel), replace with larger wattage rating resistor.

4. Defective tube or component part in erase circuit. Check in orthodox manner.

5. Accumulation of tape coating residue, dirt, dust, or foreign matter on pole pieces of erase head. Clean thoroughly. As in the case of the recording head, this procedure should be a regular part of maintenance.

Wavering of Sustained Tones. This occurrence is immediately recognized as an undesired variation of speed either in recording or playback (or both) operations.

A high-frequency variation of tape speed (usually observed as "flutter") usually indicates dirty pole pieces. A low-frequency variation of tape speed commonly is an indication of insufficient torque in the take-up motor. In this case, the motor must be replaced.

Tonal wavering may also be caused by slippage of the tape on the capstan. The capstan should be examined for roughness, out of rounding, etc. It should be cleaned often with carbon tetrachloride. Glazing is the most common fault, and is cleared by thorough cleaning of the capstan.

Erase, record, and reproduce heads should also be cleaned when this trouble occurs, as accumulation on the heads will also cause some slight variation of speeds in exceptional cases.

The drive assembly may be defective. Clean and adjust according to the manufacturer's instructions, or replace.

The forward limit switch actuator may be chattering. Adjust the actuator and limit switch according to the manufacturer's instructions.

If the tape should be binding in any part of the drive system, check and adjust or replace the following:

Oversize tape

Excessive pressure on head brake shoes

Excessive supply brake pressure

Warped supply or take-up reels (causing dragging).

Check the idler wheel for a flat spot. In some cases it may be run for 30 to 60 minutes to relieve the "permanent set." If this does not clear the flat spot, replace idler wheel.

Editing Tape

Where short intervals occur between sections of the tape to be edited (portions cut out), the tape may be pulled by hand over the playback head at approximately the correct speed for intelligibility of the sound. A little practice will do wonders here in fast and sure editing technique. As soon as the last sound of a particular sequence is heard, the tape should be immediately stopped and clipped at the spot where the tape has barely passed the playback head, unless some background sound is present that is desirable to hold a moment or two.

The correct splicing of tape is very important. An angular cut of about 45° should be made, then joined smoothly (do not overlap) together with the succeeding portion of tape. Both ends should be cut simultaneously. Splice with Scotch tape, making sure that the trim is exactly even with the width of the tape.

When it is known that the tape is to be edited, the technician should observe the following rules of good tape recording technique.

Tape Recording Technique

1. Be very careful of level fed to the tape during recording. Although modern tape of good quality is hard to "overload" under usual operating conditions, changes of average level are heavily emphasized upon editing of the tape for playback. Try to avoid unnecessary "dubbing" of a tape for purposes of readjusting respective levels of different portions of the original "take."

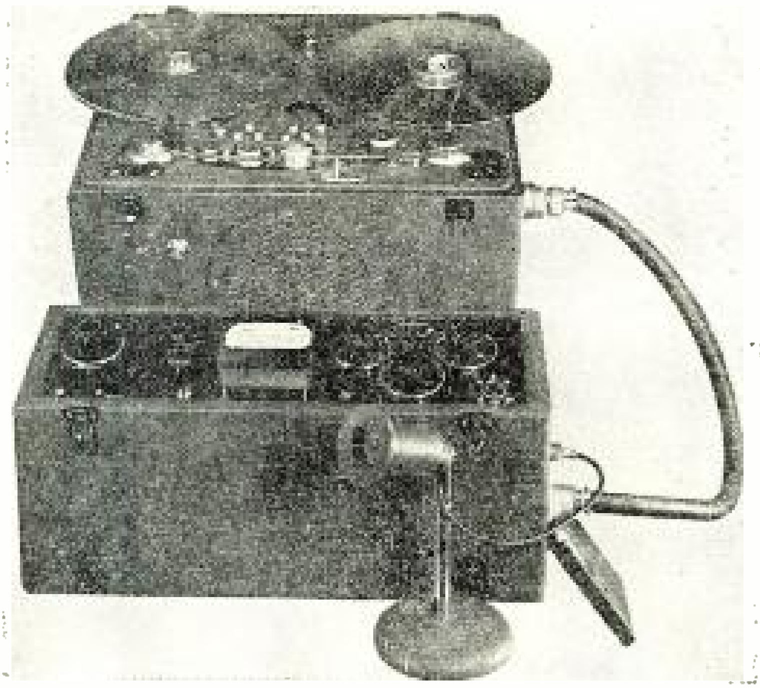
2. Watch respective levels of the background noise when tape is to be edited. An originally smooth program can be "choppy" on the air after editing when background music or noise jumps noticeably due to an omitted portion of the tape "edited out." Remember that this ratio cannot be readjusted upon dubbing.

3. For interviews, try to get some "practice" takes. The producer will often find these practice sessions more usable than the interview made for actual "take."

4. When musical numbers are to be part of a single show, and must originate from different locations under different acoustical conditions, try to avoid even slight amounts of excessive reverberation. The needed reverberation may then be added (by stations so equipped)

Left Fig. 4-36(A). Console type magnetic tape recorder. This model recorder is used in broadcast studios where high fidelity recording is necessary. In the upper portion of the console is found the amplifier, equalizer, biasing, and humbucking circuits.

Courtesy Rangertone, Inc.



Courtesy Rangertone, Inc.

Right Fig. 4-36(B). Portable type tape recorder giving high fidelity for commercial broadcast use.

upon dubbing the show. This allows control over the separate musical originations, resulting in uniform characteristics. Too much original reverberation cannot be removed.

5. Check the relative distances maintained between speaker and microphone during all recording to appear on the same program. The reason should be obvious. If the announcer finishes one sentence that was spoken two feet from the microphone, and starts the next sentence 6 inches from the mike (due perhaps to time lapse during which the announcer changed position several times, then this portion was "edited out") the listener is apt to get the impression that another person is speaking.

6. Try to educate talent and producers alike that tape recording, especially requiring editing, is a critical technique and should be so handled.

The preceding description of tape recording has been general in nature. Following is a detailed description of a specific broadcast type tape recorder, presented for the purpose of acquainting the reader with features of such equipment.

The Rangertone console type tape recorder as used in recording studios and broadcast stations is illustrated in the photo of Fig.

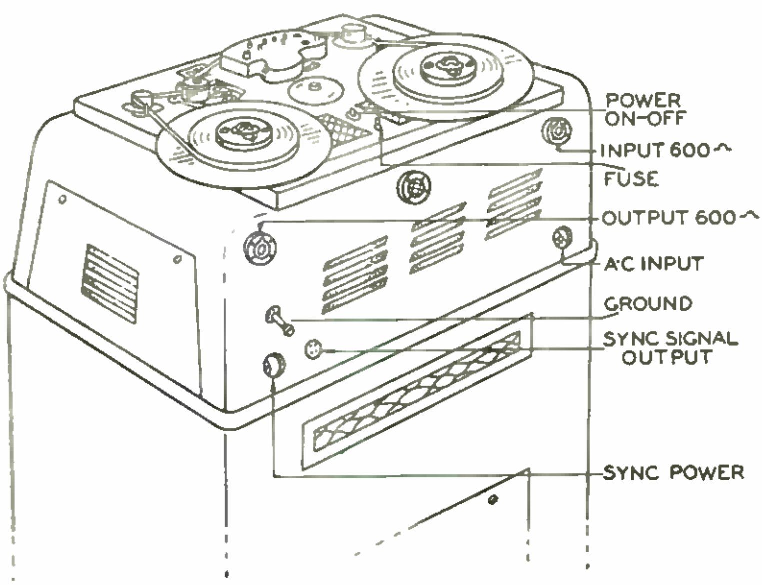
4-36(A). The portable model is shown in Fig. 4-36(B). This model is furnished with an internal loudspeaker, and a three-channel, high-gain, low-impedance microphone mixer-amplifier.

Following is the manufacturers data of the console type Rangertone tape recorder.[4]

Rangertone Magnetic Tape Recorder

General Specifications.

Input signal: Zero dbm at 600 ohms
Output signal: Minus 6 dbm at 600 ohms
Frequency band: 40 to 15,000 +2 db at 30 inches per second tape speed
Flutter: Less than 0.2 percent
Tape tension: 6 ounces
Harmonic distortion: 2 percent total
Signal-to-noise ratio: 55 db below signal at 5 percent distortion
Power Supply: Normal 110 volts 60 cps
Operation on 50 cps with special capstans
Tape speeds: 30 inches, 15 inches, and 7½ inches per second.



Courtesy Rangertone, Inc.

Fig. 4-37. Rear view of the Rangertone magnetic tape recorder, console type, showing the location of receptacles and switches.

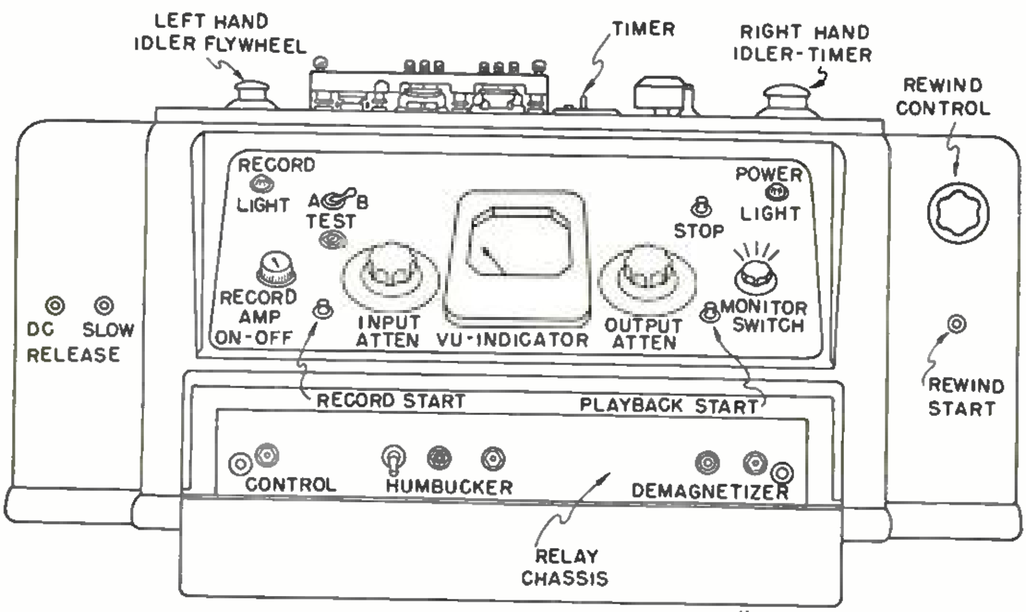
[4] Courtesy Rangertone, Inc.

Installation. Equipment should be so placed as to allow at least 8 inches of space all around it, in order to promote ventilation and cooling on continuous duty. The roller base on the studio model facilitates movement of the unit for servicing or adjustment. If desired, this base may be removed. Upon installation, it is advisable to clean the inside of the unit with a vacuum cleaner or air hose to remove all packing material. It should be cleaned similarly every few months.

The receptacle for the a-c power cord is located at the rear of the recorder (male receptacle) (Fig. 4-37). Second receptacle is also provided at the rear of the unit (female receptacle) to permit the connection of auxiliary equipment when desirable. If the unit is wired for synchronizing with film equipment this receptacle is used to supply power to the drive motor from a thyratron unit.

The three-prong female cannon plug on the right-hand side of the unit (viewed from the back) is the signal input receptacle and is wired for a 600-ohm balanced line. The cannon plug on the left is the signal output from the tape recorder and it is also wired for a 600-ohm balanced line.

Before turning on the current, an inspection should be made to see that all tubes are plugged in, and that the amplifier and power units are firmly in place. A small locking screw has been provided on the



Courtesy Rangertone, Inc.

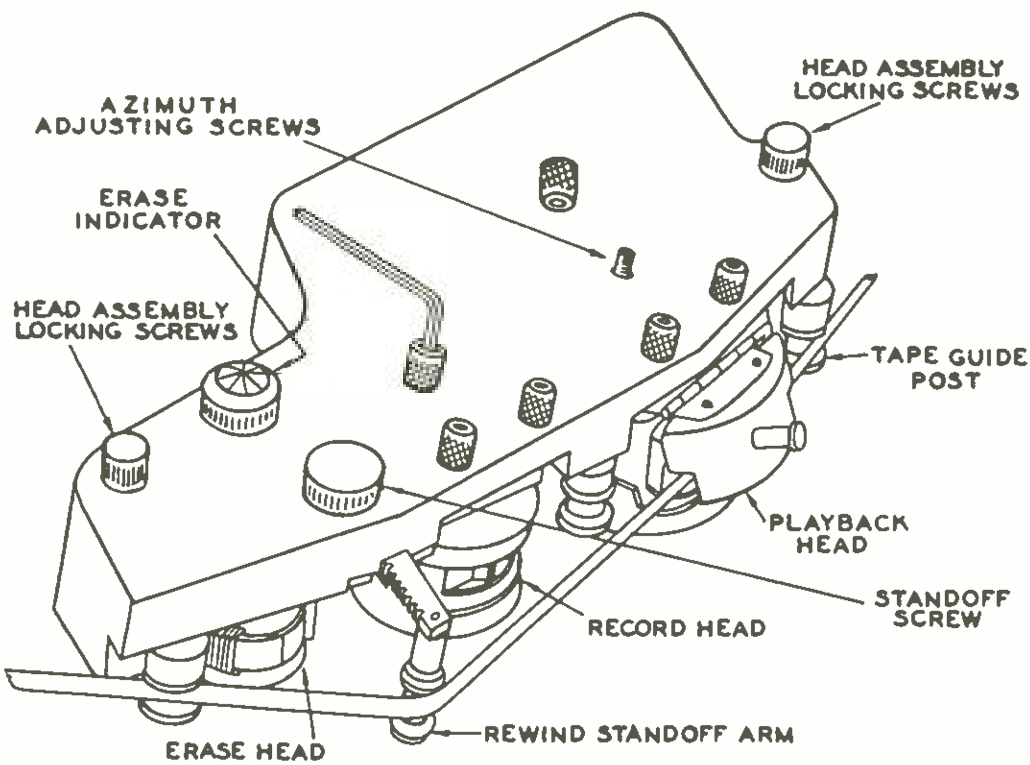
Fig. 4-38. Details of the control panel of the Rangertone console type magnetic tape recorder.

side of each unit slide, to hold the units in position once they have been seated properly.

The ON-OFF power switch is located on top of the recorder, at the rear, (Fig. 4-37) in the center of the grill. Just behind it, a 6-amp safety fuse has been provided. After the power is turned on, the green panel light to the right of the meter should come on. The red panel light to the left of the meter should light when the record amplifier is turned on. The panel drawing is shown in Fig. 4-38.

An empty plate and hub should be placed on the right spindle, by pulling the catch up and rotating it a quarter of a turn which will lock the hub in place. Similarly, a plate should be placed upon the left spindle and upon this a reel of recording tape should then be placed.

The recording tape should be unwinding in a counterclockwise direction. The side of the tape with the dull finish is the magnetic-coated side and should be on the inside of the turns. The tape should be placed around the left idler, then passed in front of the erase-record-playback heads. (See Fig. 4-39.) The small shutter just in front of the playback head is opened momentarily to permit the tape

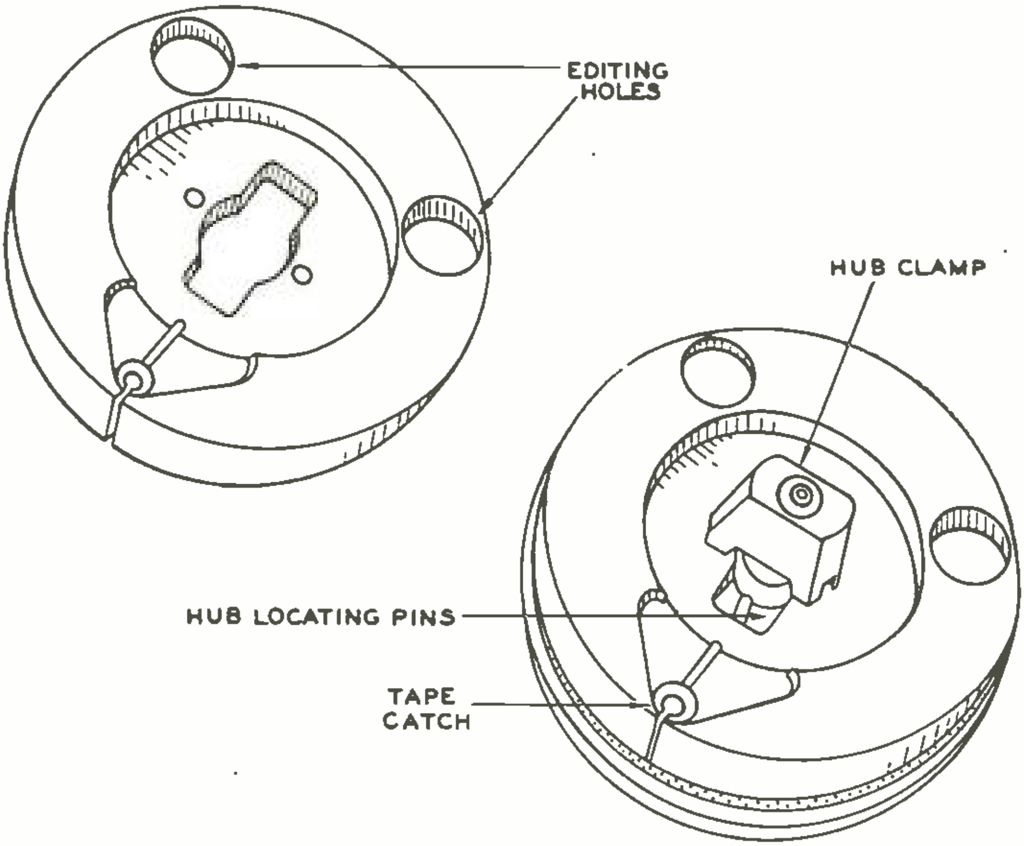


Courtesy Rangertone, Inc.

Fig. 4-39. Magnetic tape recorder head assembly including the erase, record, and playback heads, and showing the path of the tape.

to pass behind it. The tape is then passed between the motor spindle and rubber idler wheel, around the right idler wheel, over the plate on the right spindle and finally, is fastened to the right hub by means of the little catch and lever in the rim of the hub. The hub details are shown in the drawing of Fig. 4-40.

The motor on the left is the rewind motor; the one on the right the take-up motor. The synchronous or drive motor, which holds the tape to constant speed, is the motor at the front right, against which the rubber idler bears (Fig. 4-38).



Courtesy Rangertone, Inc.

Fig. 4-40. Hub assembly used to hold the magnetic recording tape on the reels of the Rangertone magnetic tape recorder.

There is no braking current on the motors until some operating relay is energized which is accomplished by pushing any of the control buttons except the "slow" or "release" buttons.

When ready to operate, it is advisable to apply braking current to the motors, so that the tape will be held firmly in position. This may be done by pushing the "stop" button, and then tightening the tape by turning either of the disks by hand. The braking may be released at any time by pushing the "release" button, located on the left side of the cabinet front (Fig. 4-38).

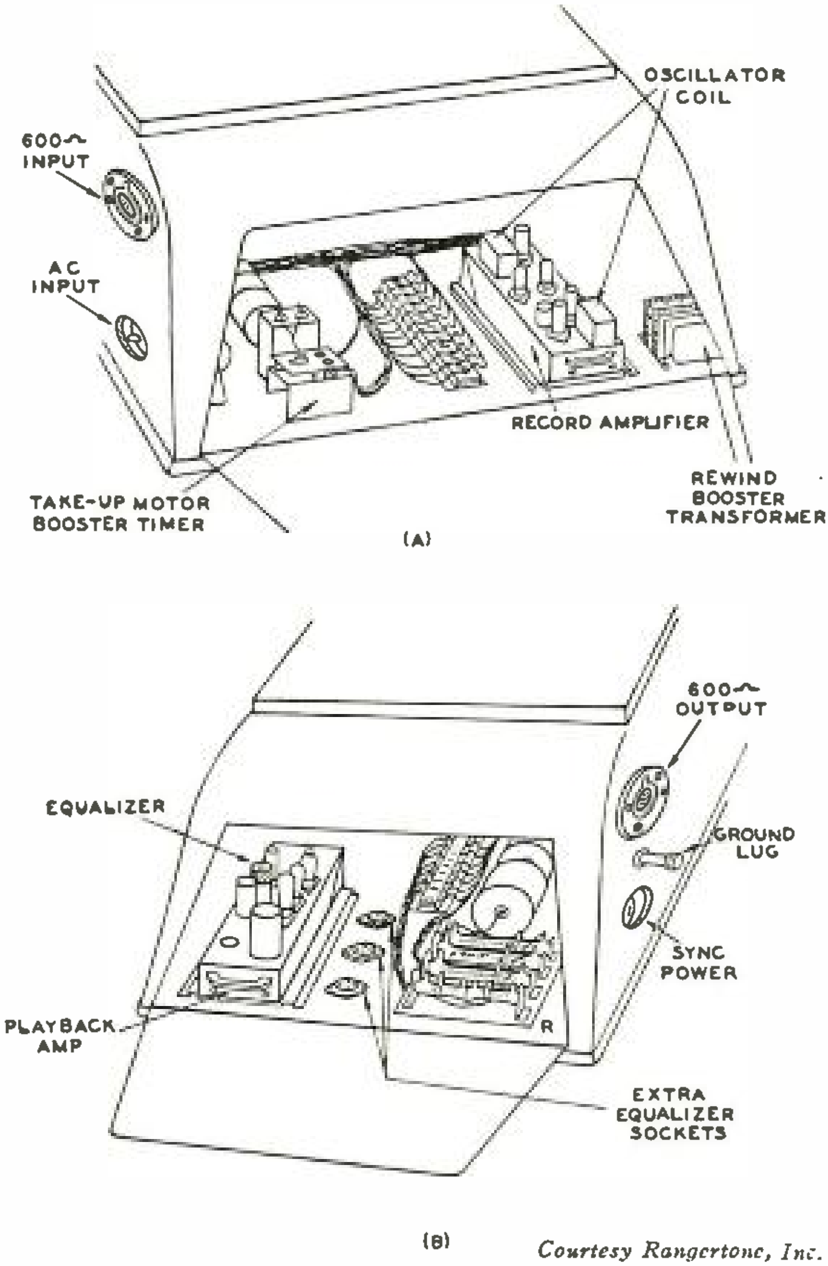
The "minutes indicator" should be set at zero before starting a recording. At the 15-inch speed, this indicator reads correctly in minutes and seconds. When operating at 30-inch speed, the reading should be halved to obtain the correct elapsed time. At the 7½-inch speed, the reading should, of course, be doubled.

A selector switch on the right of the meter panel, connects the vu meter across various voltages of the unit, to enable a check of their operation. The values indicated are, however, only relative. This switch is designated "monitor switch" in Fig. 4-38.

To Record. Turn on the record amplifier power switch (red light comes on) and allow twenty seconds for heating up. When the record amplifier has had opportunity to heat, push the "playback" button to set the tape in motion. When the "record" button is pressed, the red light (Fig. 4-39) on the head assembly should light, indicating that erase current is passing through the erase head. This second red light serves as a warning to prevent the erasure of valuable recordings. It is best never to turn on the record amplifier power switch, unless a recording is actually to be made. It should further be noted that the intensity of this red light is a good relative indication that the erase current is of good power.

If the equipment is in the braking position, the "record" button will not stay locked after pressing. This is a further precaution to prevent anything but a temporary erasing action. If it is desired to lock in the record relay, first press the "release" button and then the "record" button.

The value of the input signal may now be read on the vu meter by selecting the proper meter switch position. With "record" locked-in, the bias voltage may be determined. This is the amount of bias voltage at 90 kc applied to the recording head along with the amplified audio current. It varies with tapes but 0 vu is a good value to start with. The value of this bias may be adjusted by the screw-driver adjustment on the left of the relay box in the paneled section below the main control panel (Fig. 4-38 marked "Control"). A switch has been provided on the record amplifier, (Fig. 4-41(A)), itself, just below its handle, which may be used to obtain large bias values for high coercive force tapes. When this switch is in the high bias position, the bias control on the relay box has no effect.



Courtesy Rangertone, Inc.

Fig. 4-41. In (A) is shown the right side door of the Rangertone console model tape recorder open with the circuits available for adjustment or repair. (B), the right side of the console showing the playback amplifier and the humbucking potentiometer, *R*.

To prevent the recording of a "swish" at the beginning, note the setting of the recorder gain control (input attenuator) at the left of the meter, then turn it to minimum before starting the equipment. Push the "stop" button to turn on the braking current, then set the tape in motion by pushing the "play" button, and finally after a delay of one second to allow the tape to come to full speed, push the "record" button. The record gain control is then brought up gradually to the previously indicated position. The vu monitor switch should now be set on "output" and the signal actually on the tape may be noted.

Playback. For "playback" the black button "play" at the lower right of the meter panel should be pushed. This sets the tape in motion. To control the output, the gain control to the right of the meter may be used. The vu meter may be connected in the output circuit of the playback amplifier by turning the monitor switch to "output." This gives an indication of the playback amplifier output and not the signal delivered to the equipment's output terminals. The reason for this is to be sure that the level on the tape has the proper value, regardless of what portion of it is delivered to subsequent equipment. The meter should normally be kept in this position, either for playback or recording, as this reading determines at a glance that the entire operation is normal. The signal reading should be kept below −3 vu. As this meter circuit "loads" the playback amplifier to some extent, it is best to leave the monitor switch in one position during playback, and not to switch it on or off so as not to disturb the level. This precaution may be discounted, if the meter is equipped with a cathode follower.

Rewind. When the recording is completed it is necessary to rewind the tape back on the "left" reel. To accomplish this, push the button labelled "rewind" located just below the large control knob at the extreme right of the console. This places both the take-up and rewind motors under the control of the speed control knob above the rewind button. By rotating this control, the tape can be made to move fast or slow to the right or left.

In rewinding, the tape is moved to the left, onto the rewind motor spindle. To reduce the wear on tape and heads, it is suggested that the small stand-off knob between the erase and record heads should be turned clockwise which causes a lever to move forward that takes the tape completely away from the heads. Care should be taken to return this to the retracted position before starting subsequent recordings. Rewinding in either direction may be brought to a halt quickly and safely at any time by pushing the "stop" button, or it may be slowed down gradually by turning back the large rewind control knob.

A nonmagnetic steel clip should be placed on the round roll near the tape end to hold it in position.

Handling the Tape. While the tape stacks nicely, it is, of course, not a solid roll. Care must be exercised not to push the roll apart. The hub and the immediate turns around it are perhaps the most sensitive sections. If the tape is handled slowly at all times no difficulty should be experienced. Rolls should be carried around on disks. The roll may then be slid off onto the table or the cardboard container.

When placing a roll and the disk on the recorder, the position of the spindle catch should be noted, and the roll turned to match properly. The disk and tape should then be lowered carefully onto the spindle, and the disk may be moved back and forth slightly to ensure that all seat properly on the spindle. The hub should never be permitted to be pushed up with respect to the rest of the roll.

If the tape has been subjected to excessive changes in temperature, it may be particularly difficult to handle. Look for looseness in the roll of tape or nonconcentricity of the turns. To relax the tape under these circumstances, it should be placed on the machine carefully, threaded through the capstans and heads, and then wound on the right reel, by pressing the "rewind" button and turning the rewind control to the right. Care should be exercised not to rewind too rapidly. After the tape has been wound on the right disk, it should be rewound back onto the left. After that it may be used in the normal manner.

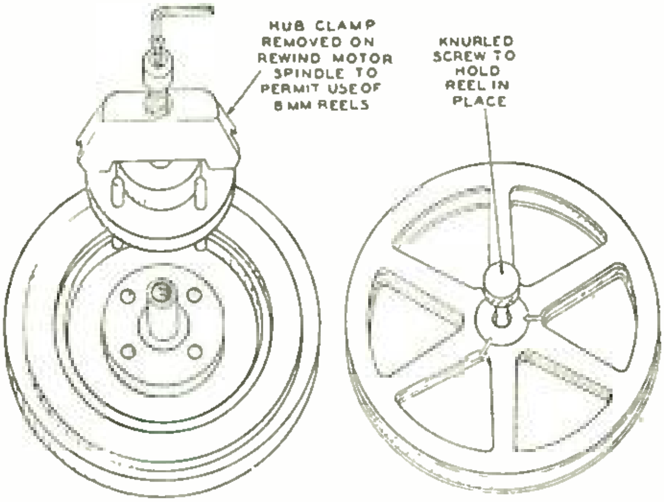
Unless kept sealed or in a humidifier, acetate base tapes will eventually dry out. The vinyl type tapes will keep indefinitely without special care, and this also applies to paper tapes. Apparently the magnetic coating on these tapes does not deteriorate, as it is a form of rust, the end product of all iron.

It takes an extremely powerful magnetic field to damage the recording on a tape. Therefore no concern over the presence of normal electrical equipment is necessary. However, the tape should be kept away from strong electric fields and excessive temperature changes.

Light yellow tapered hubs of small diameter are furnished which will fit in the center of rolls of tape supplied by the factory. These should only be used for the initial run with new tape. The tape should then be rewound on the normal and larger hubs provided with the tape catch.

To accommodate 8 mm motion picture type reels, the left reel catch may be removed by unscrewing the center Allen screw. The knob screw stored in the center of the top plate may then be used to hold the reel in place on the left (Fig. 4-42). The normal hub will still be used on the right side under these conditions.

Fig. 4-42. Method for removing the left reel hub clamp with an Allen wrench. This must be done to accommodate an 8mm reel.



Courtesy Rangertone, Inc.

The Control Unit. The control unit located behind the blank panel just below the meter panel (Fig. 4-38), may be considered the nerve center of the Rangertone magnetic tape recorder. Besides containing the relays and associated interlocking circuits, it houses three important elements: the bias control, the humbucker, and the demagnetizer.

Fig. 4-41 is a pictorial representation of this unit. Great care has been exercised to provide a foolproof interlocking network which promotes proper functioning. The various functions will now be described:

Power: To operate the equipment, first turn on the ON-OFF switch at the rear of the top plate. This provides the amplifier power supply with a.c. and the relays with d.c. for the braking of the motors.

Playback: This push button energizes the playback relay which places an a-c voltage on the rubber drive solenoid, on the hysteresis motor and on the take-up motor. At the same time a d-c voltage is supplied to the rewind motor for braking tension.

Stop: To stop the machine, push the "stop" button.

Record: This relay when operated, closes the high-frequency current to the erase head, and completes the circuit to the record head.

Rewind: The Rangertone rewinding circuit is one of the outstanding features, as its rewind control permits motion of the tape in either direction across the heads and at all speeds. Circuitwise the rewind push button energizes the rewind relay, which places a.c. on the take-up and rewind motors through two diametrically opposed rheostats operating from a single shaft. Also, the rewind control serves an additional function, which has an important relation to the smooth speed compensating braking action. This results because d-c braking voltage is applied through the rewind rheostat to the motors in the stop position. With the control turned completely clockwise so that the take-up motor is spooling, pressing the "stop" button puts full braking voltage on the opposite or rewind motor. When the rewind motor is spooling energizing the "stop" relay, it places a lower braking voltage on the rewind motor which permits the greater strength of braking d.c. in the take-up motor to override the other.

Release: The "release" push button at the front left side of the console removes the d-c braking from the motor windings. This is useful when threading or editing for in this position it is not necessary to work against the braking voltage of the motors. Furthermore, it permits energizing the record amplifier when adjusting record levels as previously outlined.

Slow Button: The "slow" push button on the extreme left of the console, provides an additional feature by permitting the synchronization of the two machines, or synchronizing a tape recording with a motion picture projector. It places a resistance in series with the main winding of the synchronous drive motor, and causes the motor to drop below synchronism temporarily. The motor will return to synchronism as soon as the button is released. The proper application of this push button is to hold it in for short intervals so that its action will not be too severe or discernible. To use it, the machine is started slightly ahead of the signal with which it is to be brought in step, and then the sound is slowed down until the two are "in step" or synchronization.

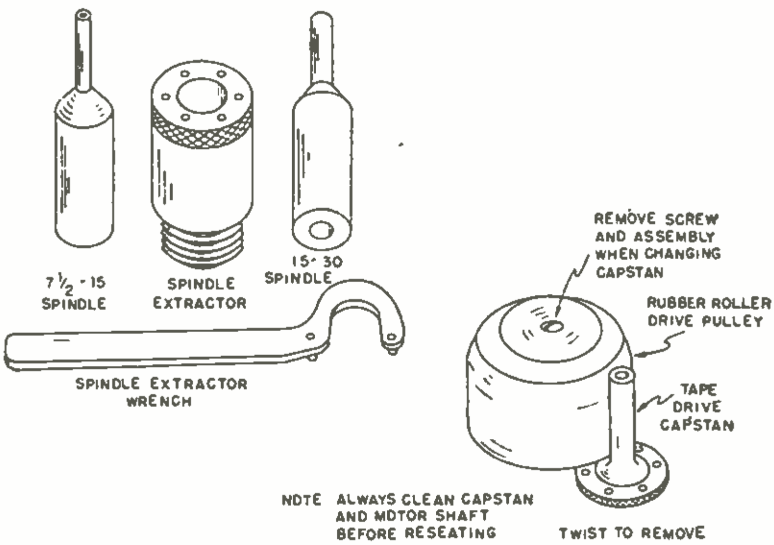
The Demagnetizer: This is a device which permits demagnetization of the record head after it has inadvertently become magnetized, as for example, after being overloaded. The need for the "demagnetizer" is indicated when excessive noise is evident in a monitored recording. It is used in the following manner, after the main ON-OFF switch and record amplifier have been turned on: Open the lower front panel, push the "demagnetizer" button, hold it in, and slowly rotate the demagnetizer "amplitude" control to its maximum (clockwise) position, and then turn it back counterclockwise all the way with the "demagnetizer" button still depressed. Repeat this operation slowly two or three times for best results. Functionally, the action of this switch and control is to place a small a-c signal on the record head and then gradually decrease it to kill any magnetism remaining in the record head.

The Humbucker: Humbucking applied to the Rangertone tape recorder is a technique for feeding an out-of-phase signal across a low resistance in the ground return of the playback head, which cancels any power hum this head may pick up as a result of stray fields. Its action is simple and, once set, is not affected appreciably by changes in line voltage, temperature, or location. Its adjustment is accomplished as follows: Do not have any tape on the machine. Connect a low-level microvoltmeter reading to −60 db to the output, and set the output attenuator at "0." Be sure that the machine is well grounded, using a good ground wire connected to the binding post at the right rear of the console (Fig. 4-41(B)). Turn the main ON-OFF switch on, and push the "stop" button. Adjust the playback heater potentiometer on the left of the playback amplifier, to give a minimum reading on the microvoltmeter. Push the "playback" button. Find the position of the "phase-reverse" switch which gives a decrease in the hum indication. Leaving the switch in this position, alternately adjust the phase and amplitude controls until the lowest hum reading is obtained.

Maintenance: Very little maintenance is required on this component other than a routine check-up of the relay contacts. To facilitate this operation, screws have been provided so that the relay chassis may be disassembled into two shells. The use of short lengths of recording tape is recommended for the cleaning of these contacts. In cleaning, care must be exercised to see that the leads within the chassis are not pushed against the moving members of the relays thus blocking their operation. It is best to remove the chassis by easing it out from both ends as well as the middle. On reinserting it, push on the far sides. A considerable amount of pressure is necessary to seat the relay chassis connector properly.

Mechanical Details. The number of mechanical adjustments has been kept to an absolute minimum, and in most cases self-contained subassemblies have been constructed to permit rapid interchange and inspection. These include the rubber-driven roller, the left-hand idler, the right-hand timer wheel, and the rewind, drive, and playback motors. Capstans of different diameter facilitate rapid tape speed change. Although these capstans are of hardened steel a slight wear is inevitable. Any appreciable eccentricity of the capstans will cause a 30-cps wow at the 1,800 rpm motor speeds. This may be checked with a standard machinist's gauge. Any capstan with an eccentricity greater than ½ thousandth may be considered as defective or worn. In some cases, this may be due to an improper setting of the tapered capstan sleeve of the drive motor. These parts should be thoroughly cleaned in naphtha or some similar solution before the tapered sections are joined. Eccentricity, remaining after these adjustments have been made, can usually be corrected by rotating the capstan to a new position on the motor shaft, to a point where such errors are cancelled out.

In seating the spindle, press it on gently with a slight twisting motion, then tap for final seating with a lightweight wooden mallet or object. To remove the spindle, first remove the rubber roller, and then with the spanner wrench provided (Fig. 4-43), turn the spindle extractor counterclockwise until the spindle pops free. Be sure to turn the spindle extractor completely clockwise before trying to seat a new spindle.



Courtesy Rangertone, Inc.

Fig. 4-43. Spindle and roller assembly and detail of the capstan drive of the Rangertone magnetic tape recorder, and the tool used for the removal of the spindle.

To change tape speed from 30 inches per second to 15 inches per second, move the toggle switch lever mounted below the top plate and accessible via the front panel door to the left. Insert the 15-inch equalizer in place of the 30-inch equalizer, on the playback amplifier and the changeover is completed. To operate with a tape speed of 7½ inches, remove the capstan as detailed above and replace with one having a diameter of 0.1592 inch.

To insure proper tension between the rubber roller and the drive spindle, a cam and solenoid adjustment has been provided. The cam shaft and the rubber roller arm are properly aligned when the two elements are almost perpendicular to each other (Fig. 4-44). This adjustment is made on the top plate by loosening the two center screws which support the rubber roller arm, and then inserting a screw-driver into the slot provided and rotating this housing about its eccentric.

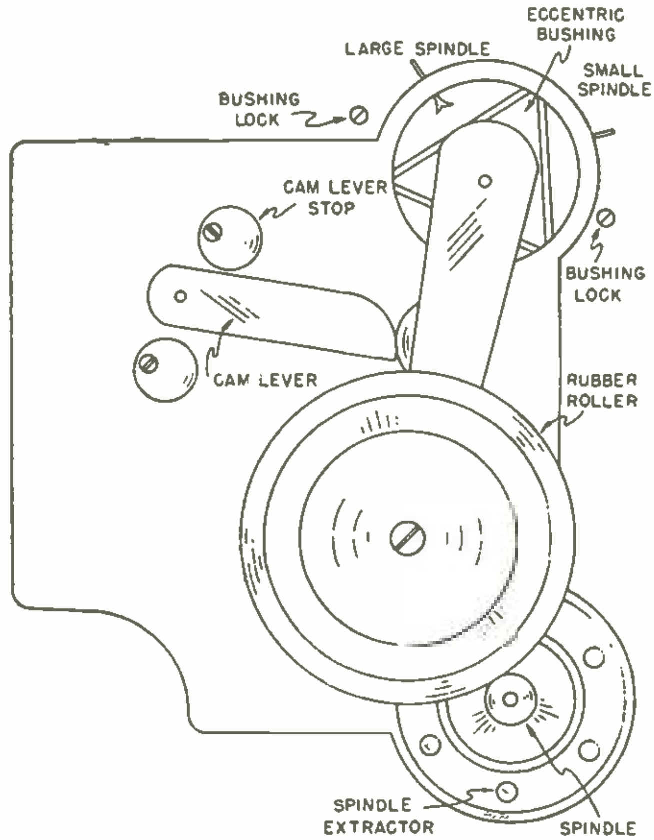


Fig. 4-44. Top view of the capstan drive of the Rangertone magnetic tape recorder.

Courtesy Rangertone, Inc.

At the 30 inches per second tape speed, misalignment in the rubber roller assembly will produce a 6 cps wow. The very finest bearings obtainable are used in this important component. Removal is accomplished by unscrewing the center supporting screw. Dirt on the rubber rim might also cause a wow and may be cleaned off periodically by applying a fine grade of emery paper to its surface when it is rotating. Any wow introduced by misalignment of the right- or left-hand idlers will occur at the rate of about 5 cps. These units are sub-assemblies. To remove the left-hand idler, remove the top plate, turn it upside down and remove the flywheel from the shaft. Next, loosen the ring pressing against the rubber shock mount, by tapping it in a counterclockwise direction, with a drive pin inserted into the hole. The right-hand idler is removed in a similar manner except that the timer worm support should be loosened before removal is attempted.

On changing capstans, it is necessary to change the cam adjustment. The various capstan diameters and related speeds are as follows:

Capstan Diameter in Inches	Tape Speed in Inches at			
	60 cps		50 cps	
	high	low	high	low
0.4584			36	18
0.3820			30	15
0.3183	30	15		
0.1910			15	7½
0.1592	15	7½		

Only capstans for 60-cps or 50-cps current are furnished with a recorder unless especially ordered.

General Care and Maintenance. For best results daily routine maintenance is suggested. With some tapes it may be necessary to clean the heads as often as once every four hours. This is readily done by using a swab saturated with carbontetrachloride. At the same time, the scraper arm resting on the drive spindle should be loosened from its normal position and any lint or dirt on the felt should be removed.

Lubrication: The drive motor will not require lubrication.

The fan should be lubricated once every six months using a good grade of light oil.

All other bearings are sealed with a lifetime lubricant.

A dab of heavy grease on the worm drive and timer parts is in order about every six months.

Removal of Top Plate and Components: The top plate may be removed in the following manner:

1. Remove the 5 Allen screws located around the edge of the top plate.

2. Remove the head assembly, and then take out the two screws from the connecting plugs for the heads.

3. Separate the connection block which supplies power to the top plate components. This may be reached from the inside of the cabinet and is located at the rear.

4. The top plate may now be removed. Extreme care should be exercised in handling the unit, as any knock or blow to the rotor shaft or idler pulleys will cause misalignment and consequent wow.

Suggestions for Best Results: When recording, a fairly good signal is best, provided that it does not overload. With the vu meter switched in the playback position, this can be measured as indicated by the red scale. The pointer should not go beyond the red scale "0" position.

Patching of the tape should be done by clipping the tape ends on a bias and joining the tape using Scotch tape applied to the nonmagnetic back side of the recording tape.

A 1,000-cps test tone is very useful for an over-all quick check. The bias may be adjusted until a clean sound is obtained. The presence of any persistent wow should be noted and the offending rotary member should be corrected.

Irregularities in the cutting of the tape occasionally give rise to variations which can only be corrected by obtaining better cut tape.

In operating Rangertone magnetic tape recording equipment, slow, calm movements are recommended at all times. It will be well to keep an eye on the various moving parts, especially the tape standoff between the record and erase heads to be certain that all is in order for that particular operation. It is advisable to return the "rewind" knob to center position with its white mark up, after each use.

Practice will quickly establish the proper co-ordination for most satisfactory high-quality magnetic tape recording.

Chapter 5

STUDIO AND CONTROL ROOM

Emergencies and Maintenance

It is a "sad but true" fact that in spite of the most thorough maintenance schedules on broadcast equipment, trouble may occur at any time. A well-rounded, consistent maintenance schedule will greatly cut the probability of trouble during the broadcast day. When emergencies do develop, the experience of the operator and his familiarity with the equipment will determine time lost.

To this we might well add a third factor—the Human Factor. Perhaps we should class this as a normal sub-part of experience, yet many times the supposedly "experienced" operator will lose as much time under pressure as a comparatively new man. This human factor is the art of being *mentally prepared* at all times for emergency procedures.

It is obvious that experience and familiarity with specific equipment cannot be gained to a great degree by reading this text. Our purpose here is to make suggestions relative to a majority of troubles and illustrate specific examples of applications of emergency procedures. We hope to accomplish this in such a way as to get YOU *mentally prepared*. We are concerned, in this chapter, with the control room. Methods of meeting emergencies, at the transmitter, are studied in detail in Part 5, Chapter 15.

In order that the following discussion be as applicable as possible for the particular type of installation with which the reader is concerned, it is necessary to divide the analysis into two general categories; the simple consolette class without associated patch-panels, and the more complex installation with means of patching inputs and outputs of the equipment.

Meeting Emergencies Without Patch-Panels

Contrary to what the student or new operator might think, the so-called "simple" type of control room, where only a control console is used without auxiliary amplifiers and patch-boards, is the most difficult type to return to the air on short notice in case of major trouble.

Some consolettes, such as the RCA 76-B5, have a means of switching the entire program circuit to the monitor or spare amplifier, and of feeding this amplifier to the regular program line. We will discuss this type of installation first.

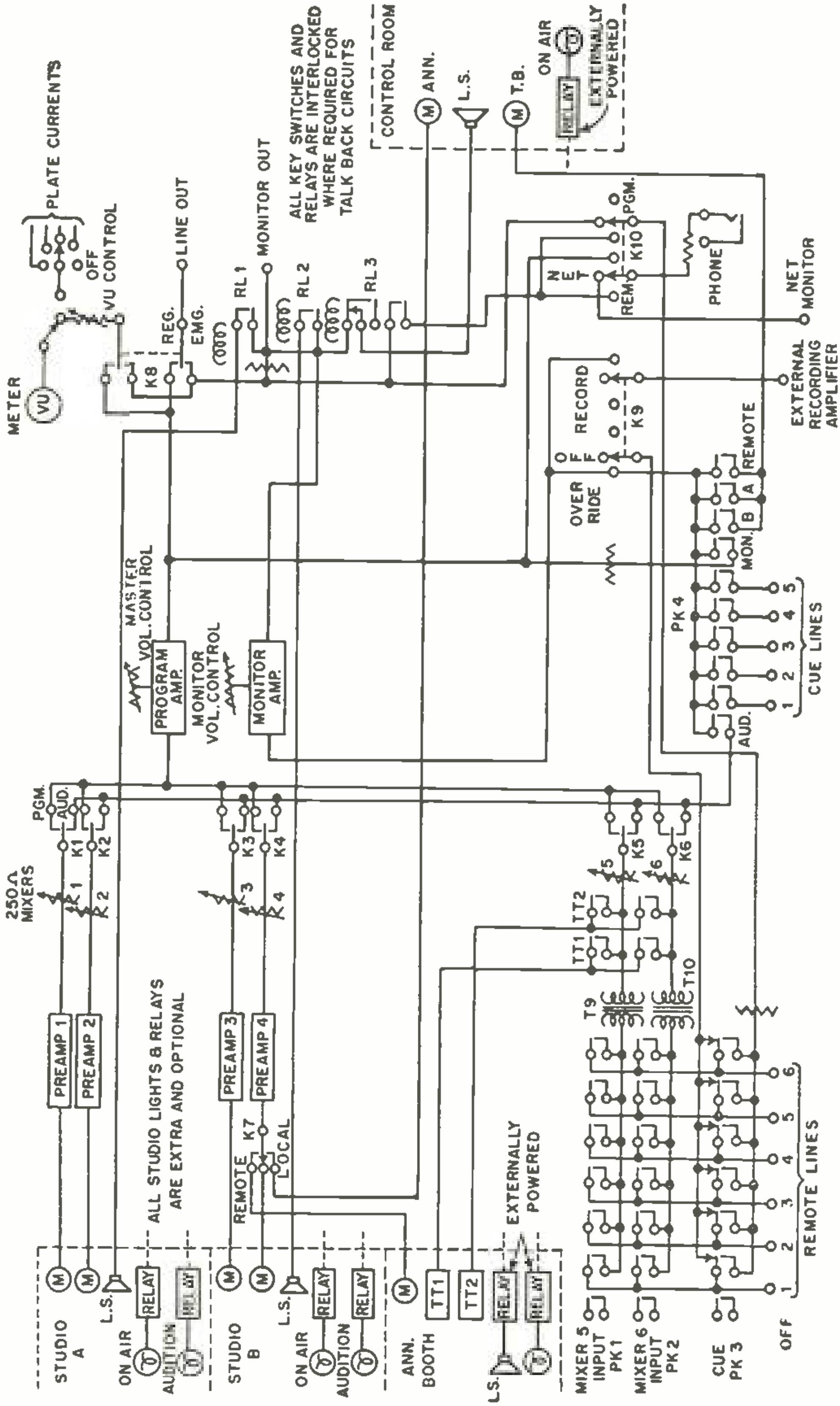
Fig. 5-1 is a simplified block diagram of the 76-B5 consolette. Every operator on the job should be able to draw such a block diagram of his particular installation. We will assume that this equipment is installed as is, without any additional patch-panels or auxiliary apparatus. Further assume that we are using the mikes in Studio A; it makes no difference at present whether we are "on-air" or on audition.

One mike channel goes dead. If we have just switched this mike "on," it is well to quickly double check for proper switching positions. If the channel dies while in use, we are obviously in trouble.

The first check is to turn the other mike "on" if it is not already hot. If this mike picks up the sound, our program channel is OK and trouble lies in the mike or cable itself or more likely, in the associated preamp. If an audition, we can take time to check the preamp tube, or simply instruct the persons to use the other mike. If the program is being aired, the other mike must be properly placed or persons moved, as quickly as possible.

Suppose, however, that both mike channels are dead. It is very unlikely that both mikes or cable would be at fault, provided they are plugged in! (Importance of rehearsals or preair mike checks.) This specific consolette uses a dual power supply; one for the program amplifier and preamps, one for the monitor amplifier. It is also unlikely that both preamps would go bad at the same time. Therefore, the most likely points of trouble are program amplifier (if we are on air) or its power supply, or in the monitor amplifier (if on audition).

If the program is being aired, our duty is to get it back in the shortest possible time. The 76-B5 has an emergency provision to supply the preamps with power from the monitor amplifier power supply by simply throwing a switch from "REG." to "EMG." position on the power-supply panel. The "line out" switch may then be thrown from "REG." to "EMG." position which connects the outgoing line to the monitor amplifier (see Fig. 5-1). By throwing the mike keys to the "audition" position, we are back on-the-air thanks to this type of emergency provision. All operating personnel should be required to



Courtesy RCA

Fig. 5-1. Block diagram of a control console such as is found in the control room of a broadcast station.

practice this procedure before or after broadcast hours when such a setup is provided.

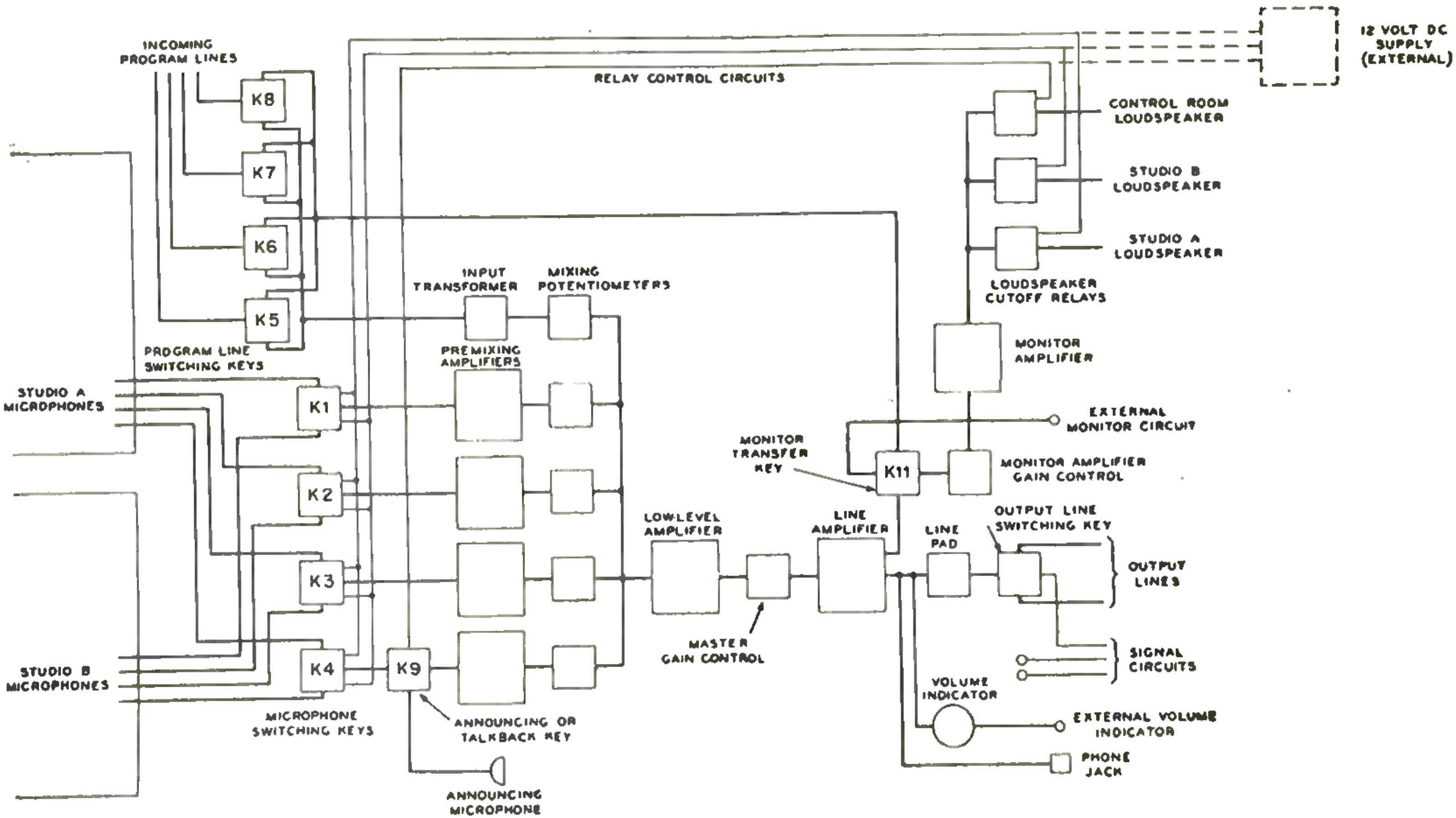
Referring to Fig. 5-1, it will be noted that the turntables (*TT1* and *TT2*) are connected through their respective keys and faders directly to the program amplifier (or monitor amplifier). This is also true of remotes and incoming network lines. Turntables frequently have preamps of their own installed in the turntable cabinets, and these should not be forgotten in case of a "dead turntable." If both pickups are silent, our obvious procedure is to carry out the aforementioned emergency precaution. This by-passes any likely trouble in power-supply or regular program amplifier. Unless the preamps in the turntable are supplied from the regular power supply, the "emergency" switch on the power supply need not be used since we are not using preamps.

In case of incoming remote or network line failure, headphones are usually provided to monitor the line as it comes in. If such is not possible, headphones should be placed across the proper terminal pair to see if the program is "coming in." If so, it is again obvious that the same emergency procedure is necessary, without the power-supply emergency switching.

All of which simmers down to this: know your equipment, plan every possible emergency checks and procedures with the provisions at hand. This is the art of being mentally prepared. It is also the secret of avoiding panic and confusion which cuts down efficiency in meeting emergencies.

The Consolette-Period. The simplest type of installation (from an initial cost standpoint) is the single-channel consolette with no emergency provision incorporated in the circuit and no associated patch-board or auxiliary equipment. Obviously, emergency procedure may take somewhat longer. So long as we meet it with the highest efficiency possible, we are doing our best. This type of control equipment may very conveniently incorporate a patch-board and added emergency equipment; we are concerned now, however, with the control room consisting of a consolette-period.

Fig. 5-2 is a block diagram of the Western Electric 23-C consolette when not used with an auxiliary patch-panel. Microphone input circuits consist of the switching keys *K1, K2, K3, K4,* and *K9* with four single-stage preamps terminating at the mixing potentiometers. Keys *K1* to *K4* select any of four mikes in either of two studios ("up" position for one studio, "down" position for corresponding mike num-



Courtesy Western Electric

Fig. 5-2. Block diagram of a control consolette used without an auxiliary patch panel and amplifiers. Making emergency repairs on such a piece of equipment demands rapid standardized procedures because of the absence of an auxiliary channel for broadcasting.

ber in other studio). Key *K9* connects an announcing or "talkback" mike in the control room to preamp No. 4 in place of the other microphones. Keys *K5* to *K8* connect any one of four incoming program lines (or high-level phono pickups or phono preamp outputs) to a mixing potentiometer or to the monitor amplifier for preliminary monitoring.

Now let's go back over comparable troubles with this type of installation as we did previously. Assume a microphone has gone dead. If the other mikes in the studio are still "live," our trouble obviously is in the microphone, mike cord, or preamp associated with that particular channel. If all microphones are dead, we may be in trouble. *Mental preparedness pays off!*

Naturally, if all the mikes are dead, we must look to a point common to all mike channels. Where is our first common point in this example? It is the single-stage low-level amplifier between the outputs of the mike mixers and the master gain control. The operator should know where this tube is and be able to change it rapidly, since the great majority of troubles occur in the vacuum tube. The Western Electric 23-C also has a meter selector switch and associated jack for the purpose of measuring cathode currents of each tube on an external meter. This tube is No. 5 on the selector switch. Obviously, if all readings are zero, trouble is in the power supply, and the 83V rectifier tube should be changed immediately. With consoles such as this, that use all glass type tubes, it is often quicker to lift the top cover and glance at the tubes, looking for a burned-out filament. If the trouble is not in the power supply or the low-level amplifier, it is in the program amplifier consisting of a 1603 and a 42 output tube.

Assume now that all tubes are OK, indicating some trouble in a component part of a circuit common to all mike channels. Unless the trouble is a complete loss of power due to power-plant failure or blown fuse on the console, (indicated by all filaments being out) it is necessary to use the external meter and jack provision to measure all cathode currents. This generally, (not always) provides a quick clue to the faulty stage. If there is simply a short or open in the signal circuit which would not influence d-c meter readings, we are really in trouble calling for further emergency measures.

Since we are discussing, at present, the way of efficiently meeting emergencies rather than actual repair of equipment, we will consider getting back on the air as quickly as possible without completely shutting down and starting a point-to-point service routine. For example, do the turntables feed into a key *K5* to *K8*, and is this channel OK? Such is possible, since in this specific instance a separate filter section of the common power supply is used to supply the preamps. If trouble should develop here (unless a short) it is still possible to use sources *K5* to *K8* since this bypasses the preamps. Recordings may then be played until trouble is cleared or further emergency provisions are instigated. If affiliated with a network, the incoming net line may be jumpered over to the transmitter line on the line terminal board. Keep clip leads handy!

If every source available on the consolette is dead and it is apparent that some time will be needed to clear the fault, there is still a last means available at nearly all stations to quickly get back "on-the-air," even though under severe limitations. This is by use of a "remote control" amplifier, of the type taken to remote points for broadcast purposes. Studio mikes may be connected to this amplifier, and the output fed to the transmitter line, or to the spare line. Every operator should know where the line terminal board is, and it is very consoling to have special clip leads available to clip the output of a remote-control amplifier to the correct line terminals. Certainly every broadcast station, no matter how small and economically limited, should provide this much of an emergency provision.

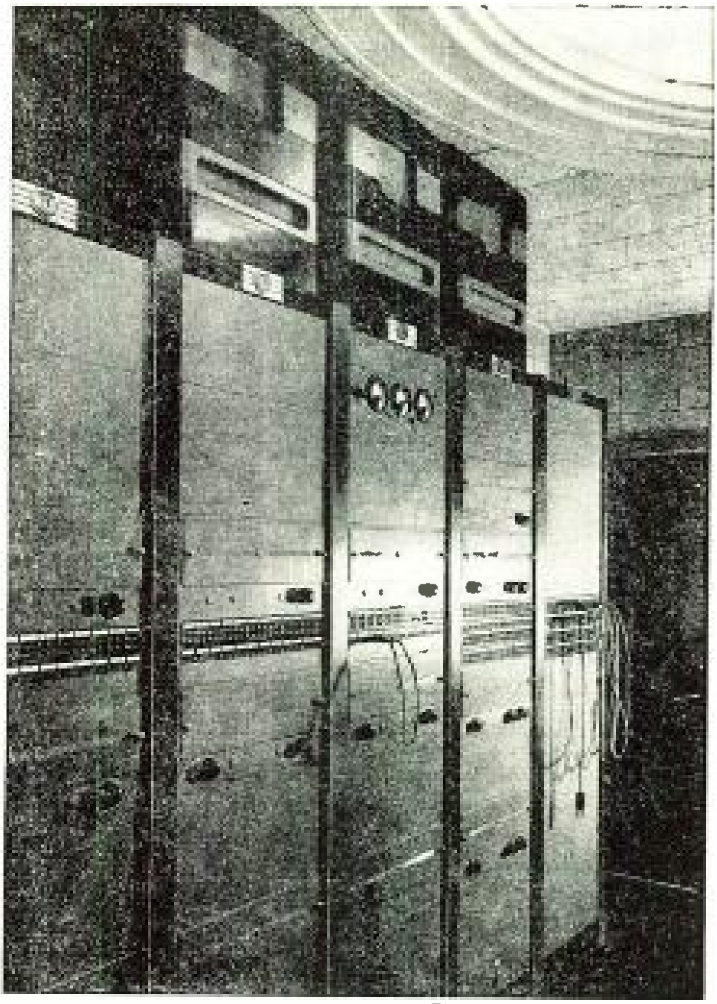
Hints for tracing down trouble in control equipment are presented later in this chapter under Maintenance.

Patch up Your Troubles

Larger studios using more auxiliary equipment and patching panels are much more efficiently arranged for emergency provisions than those types discussed above. Due to this very complexity, however, such an arrangement is apt to be very confusing especially to the newcomer. The first rule is to become as familiar as possible with the general type of larger station setup.

Fig. 5-3 is a photo of a typical amplifier rack, patch-panel installation. These are the racks associated with the block diagram of Fig. 1-1(A) and the control board shown in Fig. 1-1(B).

The rack closest to the viewer in Fig. 5-3 houses all preamps, low-level and high-level amplifiers used for Studio A. The second rack is the same for Studio B, the third for the transcription and announce booth. The fourth rack houses all amplifiers associated with the master control panel (program amplifiers, monitor bus feeds, etc.). The

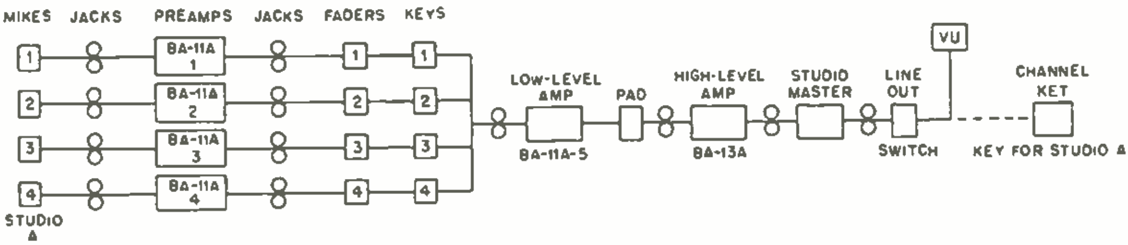


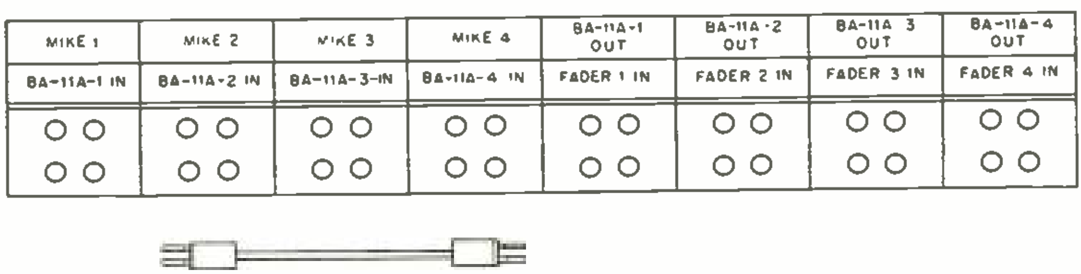
Courtesy Broadcast News

Fig. 5-3. Typical amplifier rack for a patch-panel installation. Each rack contains the preamplifiers, low-level, and high-level amplifiers for a particular studio or booth.

fifth rack is the telephone line terminations for remote control and incoming network lines.

Fig. 5-4 illustrates the complete block diagram of Studio A of Fig. 1-1(A), with corresponding jack-panel designations. Study this illustration closely to become familiar with the jacks in relation to the signal circuits. Mike *1*, for example, appears on the upper row of jacks as marked, and *BA-11A-1* preamp "in" appears on jacks just below. The following three blocks of jacks duplicate the same provision for mikes *2*, *3*, and *4*. Then follow the outputs of preamps type *BA-11A*, *1*, *2*, *3*, and *4*, and inputs to their respective faders. Then comes the jacks associated with the input and output of the low-level isolation amplifier *BA-11A* (number *5*), then the high-level amplifier type *BA-13A*, studio master fader and line out switch connections. In this installation the "line out" switch may be thrown either "down" for possible feed to the corresponding Studio A key on the master control panel, or may be placed on the "up" position which connects the studio output to a monitoring amplifier for audition purposes without tying up one of the three main program channels. Also in this particular installation, the preamps type *BA-11A* number *1* through *4*, and the isolation amplifier number *5* obtain their operating voltages from the high-level amplifier type *BA-13A*.





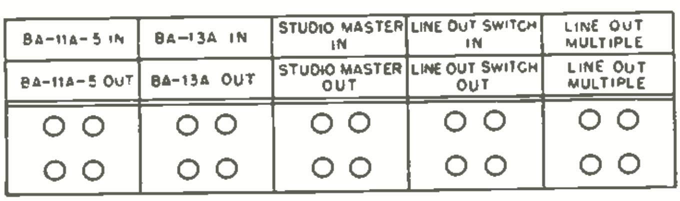


Fig. 5-4. Block diagram of control channel for studio A of Fig. 1-1(A), and jack-panel designations for each of the blocks.

It is obvious here that such flexibility is more compatible with emergency conditions than types previously discussed.

Assume, for example, that the program stops as indicated by the monitor loudspeaker and no meter swing on the master control panel vu. Since this type of installation uses a separate vu on the output of each studio panel, it provides a quick visual check as to whether the main program amplifier is at fault or the trouble is in the studio circuit itself. Should the vu indicator of the individual studio still be "swinging," it is simply necessary to push the corresponding studio key on channel 2 or channel 3 on the master control panel, and feed the output of this new program channel to the transmitter line. If the studio vu is not indicating, the trouble obviously is in the studio high-level amplifier *BA-13A*.

Where the trouble is an obvious one from the start, such as a quick visual examination of the rectifier tube in this amplifier indicating it to be dark, it is advisable to reroute the signal by means of the patch-panel. Studio B amplifier rack is an exact replica of Studio A rack and patch-panel. If we are using four mikes in Studio A, it is necessary to patch the outputs of each mike into the corresponding preamps of Studio B. For example, mike *1* of Studio A to *BA-11A-1 IN* on Studio B rack, mike *2* of Studio A to *BA-11A-2 IN* on Studio B, etc. A patch cord may then be inserted from the Studio B "Studio Master Out" jacks to the Studio A "Line Out Switch In" jacks, and we have substituted Studio B amplifiers for Studio A operation, but we must now use the faders on Studio B operating panel.

It would be possible, of course, to simply patch the mikes of Studio A over to the preamps of Studio B, then press the Studio B key on the master control panel with the "line out" switch on Studio B "down." Note, however, an important complication if this were done. When the line out switch is in the down, or program position (as distinguished from "up" or audition), and the studio key is pushed on one of the master control amplifiers, the speaker is cut off in the studio to prevent feedback. If we patch these mikes into another studio and press the corresponding key on master control for the substitute studio, the speaker in that studio will be cut but not the one in the studio "on air." This is apt to cause feedback and confusion to the performers as well as the operator. If he desires to carry out this operation without returning to Studio A "Line Out Switch" in the previous example, he must press the Studio A key or a spare channel which would cut the speaker in Studio A, or throw the line out switch of Studio A to the "up" or audition position which also cuts the speaker. It is best, however, to return to the Studio A position as in the example given to avoid confusion in getting out of the emergency operating procedure when the trouble is cleared.

Assume now that only one microphone goes dead. As always, the first thing to do, if using only one mike, is to turn another mike "on" in the studio to ascertain if the entire studio circuit is dead. If only one has failed, it is a quick and simple matter to substitute another preamp for the mike by means of patch cords. For example, if mike *1* in Studio A fails, we may patch from mike *1* to any of the preamps in Studio B, from the output of that preamp back to "Fader 1 IN" on

Studio A patch-panel. If the mike is still dead, we are faced with a microphone failure and simply must substitute another one.

In patching installations of this type it is also a simple matter to reroute network or remote signals when trouble develops in the control room. The network or remote line may be patched directly to the "transmitter line in" jacks, by-passing all studio equipment.

All of which amounts to this, know the equipment and arrangement, then *be mentally prepared.*

Studio Maintenance

Microphone Maintenance. The microphone is perhaps one of the most delicate pieces of equipment associated with broadcasting systems. Yet it is apparent to the experienced man that with careful handling the mike has outworn many sets of tubes and component electrical parts in the amplifier.

With Careful Handling. When it is necessary to transport microphones from one place to another, it is best to use a special box containing no other equipment for the purpose. The box should contain padding not only enough to take up shocks of exterior bumps, but to prevent free movement of the mike within the box. Sponge rubber seat pads are excellent to line the box with, and heavy felt material is good to wrap around the instruments and to stuff superfluous space to prevent movement.

Microphones having permanent magnets as component parts (moving coil, ribbons, etc.) should not be placed on a work bench or any place where there is a possibility of iron chips or filings being attracted to the magnet.

First Steps in Testing. It should be understood at the outset that there are a number of troubles, in modern high-quality microphones, that should be treated only at the factory of the manufacturer. It will be the purpose of this section to acquaint the reader with test procedures that will determine what to do and *what not* to do in the way of repairs.

First, of course, it is necessary to have a good audio amplifier of known characteristics, with the proper input circuit and impedance to match the microphone under test. There are three general classifications of troubles as follows:

1. No response at all
2. High noise level with or without some signal
3. No noise level but weak and perhaps distorted signal.

It should, of course, be obvious that the average technician is unable to check field response patterns or run frequency-response curves that require laboratory apparatus and soundproof rooms.

As in all troubles of microphones, it is absolutely necessary to picture in the mind the relation of the input circuit to the schematic of the mike.

For example, consider a typical high-impedance input circuit with an open circuit jack. The high-impedance mike uses the two-conductor cable, the ground or jack sleeve connection serving as the braided shield about the "hot" lead.

If the response from the mike is zero when connected to an amplifier known to be good, several possibilities occur. Either the "hot" lead is open (if the grounded side were open it would result in noise), a short exists, or the internal element of the microphone is defective. The first places to check for any defect are in the plug or the point where the cable leaves the microphone housing.

In the case of the above open circuit jack, some noise usually exists before a mike is plugged into the input. If this noise is killed upon insertion of the mike plug, but no response is obtained, a short is indicated. If the noise level remains the same or is slightly raised in intensity, an open is likely. If a closed circuit jack is used, a short will not result in lower noise level but an open will likely raise the input noise, although very slightly in some cases.

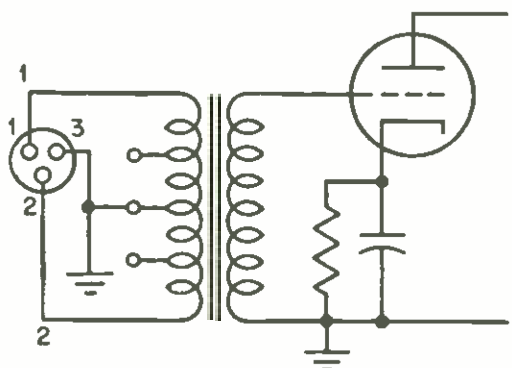


Fig. 5-5. Circuit diagram of low-impedance mike input. In this diagram pin 3 is shown grounded, actually any one of the pins may be grounded and the other pins will be "hot."

Fig. 5-5 shows a typical low-impedance mike circuit, where a three-conductor cable will be used. In this case, an open of either No. 1 or No. 2 wires will cause a "dead" microphone (so also, of course, will a short) whereas a break in the shielding will result in higher noise level. Sometimes the trouble is only intermittent and must be traced down by "jiggling" the cable starting at the lug and working back a foot or so at a time to the microphone housing. This is done by rapidly looping and straightening a small section of cable between the hands.

Following is a good general procedure to check for microphone and cable troubles.

1. Check the plug and receptacle. All types are encountered in mike input circuits. Some simple two-conductor mikes use the familiar jack and jack plug; some use a metal shell with a single pin in the center insulated from the outer conductor. The receptacle is a matching female type with the center pin being gripped tightly by a spring connector with a knurled metal ring connector for the outer conductor. The three-conductor circuits vary considerably in design but all are general insofar as inspection is concerned. Some (such as the cannon type) have a small lever on the receptacle shell which must be depressed to pull the plug from the receptacle. Other types have a small knurled knob on the shell of the plug which must be depressed. In connecting the plug to the receptacle, it is properly oriented in the receptacle and when pressure is applied one of the pins springs up through a hole in the receptacle locking the two parts together.

Plug connections that are made inside the shell require removal of the shell for inspection. In some cases the shell and plug body are both threaded, and may simply be unscrewed. Others are held together by clamps and screws. Also, some cable conductors are soldered to the pins, some are held by screws on a pin lug.

Check the connections to the pins for looseness, corrosion, dirt, faulty insulation, broken wires, or bent pins.

Check the plug body for damage, dirt, or corrosion.

Check the shell for dents, cracks, dirt, or corrosion.

While taken apart, clean everything with a cloth and cleansing fluid. Corrosion may be removed with a small strip of crocus cloth.

Check connectors of the spring type for proper contact and tension. In cases where the plug is difficult to connect or remove, coat the pins thinly with vaseline or "Lubriplate."

2. Check the microphone cable about 10 inches to a foot at a time. Loop and unloop this amount of cable between the hands while slowly twisting it at the same time. Listen to the output of the amplifier and continue to do each small section this way for at least a quarter of a minute. Broken insulation or wires will show up definitely in this way.

It should be said here that if a break is found, it is far better to replace the entire cable than to remove the faulty section of cable and splice together. Of course, this splicing may be done for emergencies when it is absolutely necessary to use the mike before an entire new cable can be obtained.

A number of microphones have a switch for turning "off" and "on" to allow greater flexibility in applications. If this switch is a sealed type with nonaccessible contacts, it should simply be checked for proper working order and if suspected, replaced. If the contacts are accessible, check in the following manner:

Inspect the terminal connections for tightness and cleanliness.

Check mounting for firmness.

While operating switch, observe all moving parts for freedom of movement, and look closely at the stationary spring contacts to ascertain their tension and good or doubtful electrical contact. Contacts that have lost tension may be tightened with fingers or pliers. Tighten all terminals possible.

Any section of the switch that is dusty, corroded, or pitted should be cleaned with a dry cloth. For more serious conditions, the cloth should be moistened with cleaning fluid and rubbed vigorously over the parts affected.

Where points of contact with the moving blade show signs of excessive wear, replace the entire switch. To clean, crocus cloth dipped in cleaning fluid may be used. For severe corrosion, #0000 or #000 sandpaper should be used, and the contacts polished clean afterward.

If dryness and binding is noticed, apply a drop of instrument oil with a toothpick to the point of motion. *Do not allow oil to flow into the electrical contacts.*

The above steps are the preliminaries to checking any faulty microphone. A great majority of the common faults are found in the receptacles, plugs, or cables. If these items have checked OK, it is necessary to proceed according to the type of microphone concerned.

The Carbon Microphone. These makes are still used in some applications such as talkback circuits and special events. Check first the power supply for the carbon buttons. If a battery is used, try a new battery. Button current should normally be 7 to 15 mils, with some cases up to 50 mils per button.

Nearly all troubles connected with the carbon mike (particularly the older models) result from a "packing" or cohering of the carbon granules, such that distortion is extremely high and output level is practically nil. A filter system will greatly reduce the chances of such carbon packing by electrical "kicks" in the carbon buttons.

It is usually possible to restore a carbon transmitter to working order by gently tapping the buttons with the fingers. Sometimes, however, the carbons become so burned that they become completely inoperative. If this happens, it is best to obtain new buttons already packed to replace the original buttons.

It is also possible to procure only the carbon granules to repack the buttons. About 0.06 cc is used in each button, (not packed full) which amounts to around 3,000 granules. The carbon granules are made usually from crushed anthracite coal with a size tolerance such that they may be sifted through a 60-mesh screen and caught by an 80-mesh screen. They are treated with hydrofluoric and hydrochloric acid, and undergo a heating process in an oven. For this reason, it is always best to buy commercial carbon granules to replace the original defective packing.

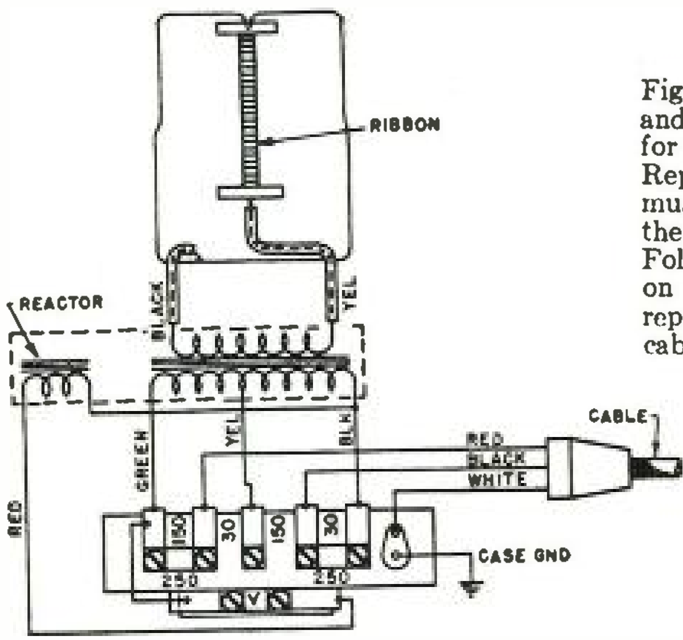


Fig. 5-6. Schematic diagram and terminal block connections for a ribbon-type microphone. Repairs on such microphones must be made carefully because the ribbon is easily damaged. Follow the color coding shown on the schematic diagram when replacing the transformer or cable.

Courtesy RCA

Crystal Microphones. Where the trouble in crystal mikes has been traced to within the unit itself, the cause will almost invariably be the crystal element. When subjected to high temperatures such as exposure to hot sunshine or similar circumstances, the unit becomes dehydrated or in a state of loss of natural moistures, and is useless. Sometimes dropping the head will cause a fracture of the crystal element. In any case, it is advisable to return the microphone to the factory for replacement of the crystal. This is a process requiring factory methods of assembly and adjustment.

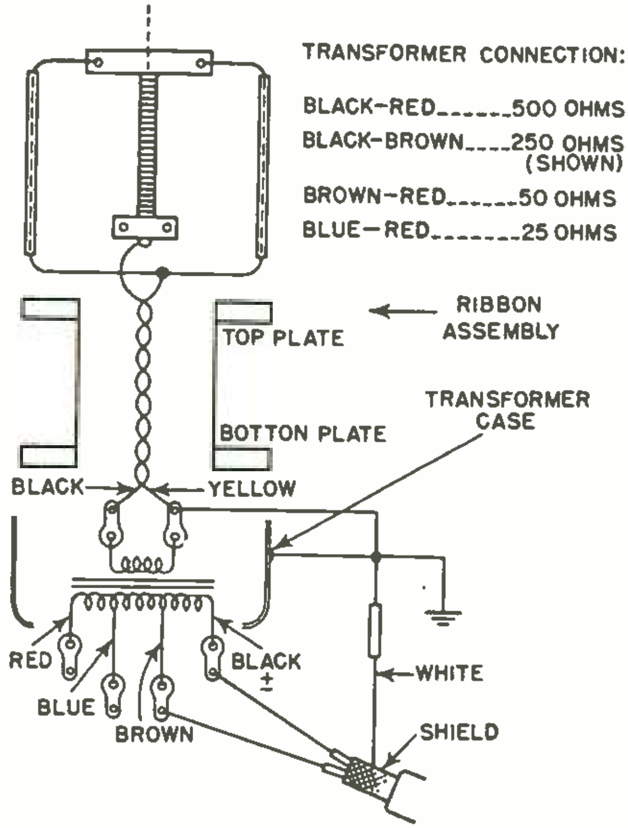


Fig. 5-7. Schematic diagram of the RCA 77B1 ribbon microphone showing the output cable connected across the 250-ohm impedance branch of the transformer. To change the output impedance of the microphone, for purposes of matching to different impedance loads, the table on the upper right side of the diagram is used. For example, to obtain an output impedance of 500 ohms, connect the output cable between the red and black terminals of the transformer.

Courtesy RCA

Ribbon and Combination Microphones. Where cables and plugs have been definitely eliminated as sources of trouble, the transformer and terminal block connections should be checked. It is very important here to remember *not* to check transformers with a battery-powered continuity meter without first removing the ribbon connections, otherwise the ribbon is apt to be damaged. *Be very careful* of applying too much heat when removing and resoldering these connections. Transformers may be replaced but if the ribbon or ribbon assembly is damaged, the microphone should be sent to the manufacturer for factory repair. Ribbons sometimes become stretched under extreme sound pressure or dropping, and consequent distortion results. Keep ribbon microphones out of extremely high winds unless they have special wind shielding as recently added to this type instrument.

Figs. 5-6, 5-7, and 5-8 show the color coding and terminal block connections for proper impedance matching of representative types of RCA microphones.

Western Electric 639 A and 639 B Microphones. (See Chapter 18 for description.) These microphones are designed to operate into an input impedance of 25 to 50 ohms, either balanced or unbalanced with respect to ground. Three pins on the microphone accommodate the 442 A jack for external connections. Pins 1 and 3 are the microphone output connections, pin 2 the ground to microphone housing. When two-conductor mike cable is used, the conductor should be connected to 1 and 3, and the shield connected to pin 2. With three-conductor cable, the third conductor is connected to pin 2 and the shield. Ground connection should not be made to pins 1 or 3 at the microphone, but either conductor may be connected to the shield and ground at the preamplifier when the input is unbalanced with respect to ground. In any event the shield should be grounded at the preamp end of the mike cable.

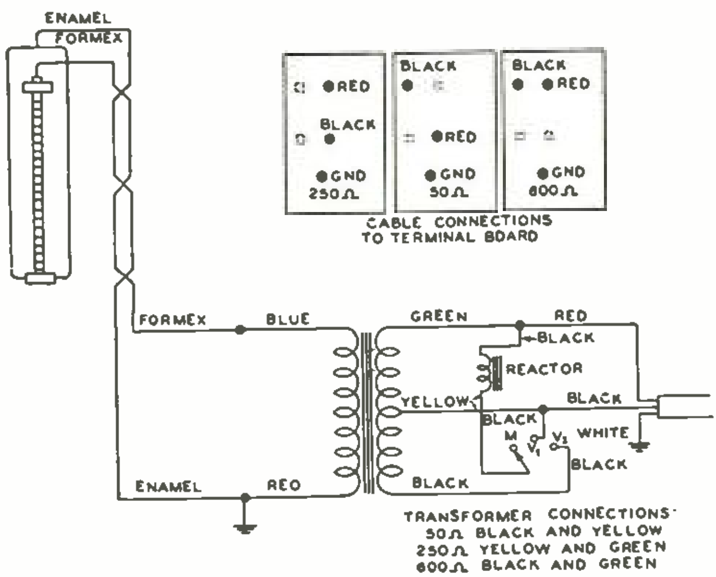


Fig. 5-8. Schematic diagram of the RCA 77D polydirectional microphone. The switch shown in the secondary circuit of the transformer may be adjusted for music (M), or voice (V_1 or V_2). This switch is adjusted by means of a screwdriver. To obtain various output impedances from the transformer, use the table of connections shown in the lower right corner of the diagram, and the connection setup shown in the upper right hand side.

Courtesy RCA

When the 639 type microphone is to be used with equipment designed to operate from higher impedance than 50 ohms, an input matching transformer such as the Western Electric 172A repeating coil should be used. This coil is a high-quality low-level matching transformer with a 30 to 250 ohms impedance ratio. This may be used with a loss of only 1 db, and will not affect the frequency characteristic of the microphone.

Phono Pickup Units and Associated Equipment. Optimum stylus tip radii for various classes of recordings are as follows:

0.003 inches (3 mils) for standard 78-rpm records

0.0025 or 0.002 inches (2.5 or 2 mils) for broadcast transcription

0.001 inches (1 mil) for LP (microgroove) records.

In actual practice, a 2.5-mil stylus is usually used in broadcast pickups for both 78-rpm records and 33⅓-rpm transcriptions, while a special 1-mil stylus must be used for the 33⅓-rpm microgroove records.

Checking Stylus Pressure. Pickup arms for broadcast use have an adjustable feature allowing control over stylus pressure against the record groove. It is very important that this amount of pressure be correctly adjusted for optimum system performance and minimum record and stylus wear. Standard record and transcription arms in high-quality systems have a rating of between 20 and 35 grams (1 ounce = 28.35 grams), depending on manufacturer. Microgroove pickups use extremely light stylus pressure of 6 to 8 grams. These pressures should be checked monthly with the manufacturers' rating.

Several good makes of special type scales for this purpose are on the market and should be a part of every maintenance department. Where such is not available, a regular postage scale may be used for *standard* type recordings. The pressure must be measured with the stylus in *playing position,* thus when using postal scales, they must be placed off the turntable and adjusted so that the stylus is at playing height when measured.

General Test of Pickup Units. When the trouble has been traced down to the pickup unit, step number one is to check the stylus. Excessive record noise and signal distortion is often caused by a defective stylus tip. Check the tip under a strong light and magnifying glass. Chips or excessive wear may be detected readily in this manner. Of course, the easiest and obviously best check is to replace the stylus with a new one if such is immediately available. In cases of permanent-type stylus assemblies, usually using diamond tips, the head must be removed and returned to the factory for stylus replacement if the tip has been damaged by dropping or other accident.

Be very sure that the stylus is secured properly in the holder. In the case of a bent shank stylus, the bend must be aligned properly with the record grooves, not turned even slightly in either direction from the center of the groove.

Check all spaces around the stylus entrance for dust or lint that often clogs the free spaces around stylus holder or pole pieces in magnetic pickups. These spaces must be cleaned thoroughly. A low-pressure air stream aids considerably in this process.

Pickup heads must be examined closely for breaks in connections, wire, or shielding. Plugs and receptacles must be gone over as suggested previously on microphone maintenance.

Where terminal boards or connecting panels are used as in some commercial installations, they should be carefully inspected for cracks. breaks, dirt, and loose connections or mountings. Each connection should be examined. Tighten all *clean* terminals, screws, lugs, and mounting bolts, being careful not to overtighten causing cracks or breakage. Any connection that is dirty, rusty, or corroded should be disconnected. Clean each part individually and thoroughly with a clean cloth or crocus cloth moistened with cleaning fluid. Replace and tighten the connection.

Nonoperative or signal-distorting crystal pickup cartridges must be replaced by new ones of the exact electrical and mechanical equivalents. There is no repair or adjustment procedure that will renew "life" to a ruined crystal element.

In the magnetic-type pickup, extreme care must be exercised in removing the cover to prevent damage to the delicate stylus and/or armature assembly. Check the centering of the armature between the pole pieces. Always check air gaps for collection of dust and lint. In many instances, transformers are located within the pickup arm. These leads and connections must be carefully checked for continuity and tightness. In some of the older-type magnetic pickups, bushings and supports that center the armature become worn and deteriorated. These replacement parts must be ordered from the manufacturer by model number and part number. It is always advisable, and often cheaper in the long run to return the pickup head to the factory for repair. *This is imperative in the case of the very-high-quality magnetic pickups such as used in broadcast and similar applications,* where a permanent-type precious metal stylus tip is employed.

Checking Frequency Response. The broadcast engineer finds it both interesting and helpful to know the actual frequency response of a given pickup unit. This information is actually necessary if complete individual equalization circuits are to be checked not only to meet the requirements of compensating a certain recording characteristic (such as preemphasis of highs as in Orthacoustic recordings) but to compensate the deficiencies of the pickup unit used. Equalization, for example, would be unnecessary (indeed harmful) should the pickup unit "roll-off" at high frequencies of its own accord!

Obviously, in order to check frequency response of a reproducer unit, the frequency run must first be made of the amplifier used. This is done in the orthodox manner using a variable frequency audio oscillator (30 to 15,000 cps) and a volume indicator both on the output of the oscillator and output of the amplifier under test. A single volume indicator could, of course, be used with a suitable switching arrangement. The single tone is fed into the amplifier at a constant level for each frequency, and the volume-indicator output meter reading noted. 1,000 cps is usually used as the reference frequency; that is, the gain is adjusted on the amplifier so that at 1,000 cps the output meter reads 0 db. Deviations at other frequencies over the range to be tested are then taken and plotted graphically.

For checking the pickup unit, major recording firms put out frequency test records that usually start out with a 1,000-cps tone to "set level," used as the reference frequency. They then jump to 10,000 cps and work down to 20 or 30 cps, a voice identifying each tone immediately prior to it. The output volume indicator on the amplifier is noted for each frequency, the gain having been adjusted to zero at 1,000 cps as with the oscillator test. This curve may then be plotted against the amplifier curve to obtain the pickup response curve. Remove any equalization associated with the pickup.

In this case, it may seem that if the amplifier response curve was perfectly flat, the pickup curve run would be the actual response of the unit. Assume, however, that at 2,000 cps the amplifier curve is +4 db, and the pickup curve is +1 db, the pickup response, therefore, is actually —3 db at this frequency, since we must subtract the 4-db gain of the amplifier at this point on the curve.

Western Electric 109-Type Reproducer Group [1]

Fig. 5-9(A) and (B) shows the Western Electric 109A and 109B pickups for high-quality commercial applications. The 109-type reproducer group is for use in electrical transcription systems for radio broadcasting and similar applications and consists of a 109-type reproducer with its supporting arm and arm rest and the necessary equalizing equipment. Each reproducer group is so constructed that it can be used for reproducing from either vertical- or lateral-cut disk-type records through the selection of the proper electrical connections by means of a switch supplied as a part of the equalizing equipment.

The 109A and 109B reproducer groups are similar except for the component 109-type reproducers which differ only in the material and

[1] Courtesy Western Electric.

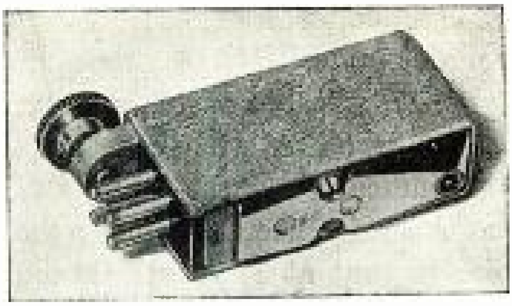


Fig. 5-9(A). Dynamic-type reproducer for both hill-and-dale records and lateral transcriptions. The diamond stylus used in this unit is especially suited for narrow-grooved, vertically-cut records.

Courtesy Western Electric

tip radius of the stylus. Both the 109A and 109B are mechanically and electrically interchangeable on the 5A arm and function equally well with the other components of the 109-type reproducers.

The 109A reproducer has a 2-mil radius diamond-tip stylus and is intended primarily for use on the transcription-type records made of nonabrasive material, with narrow- or medium-groove cross section, whereas the 109B reproducer has a 2.5-mil radius sapphire-tip stylus

Courtesy Western Electric

Fig. 5-9(B). A dynamic-type reproducer for wide-grooved, lateral-cut disc records. The major difference between this reproducer and the one shown in Fig. 5-9(A) is the stylus. The reproducer shown here uses a larger-diameter, sapphire-tip stylus.

which is designed for optimum performance when used with records having a wider-groove cross section.

The 109A and 109B reproducers are permanent-magnet, moving-coil type reproducers consisting essentially of two coils assembled on a yoke which is attached through a driving rod to a stylus. A flat suspension spring suspends the coils from a point at the center of the yoke and the coils are free to vibrate, under the influence of the stylus motion, in the field of a permanent magnet, in either a vertical motion through the magnetic field or in an oscillatory angular motion having its apex at the junction between the yoke and suspension spring. The magnetic field is so arranged that when the coils vibrate vertically, voltages of like polarity and equal amplitude are generated in each coil. When the coils are subjected to an oscillatory angular motion having its apex at the suspension spring junction, the magnetic field configuration causes voltages of opposite polarity, but equal amplitude, to be generated in each coil. Vertical-cut recordings result in vertical coil motion and the two coils are connected by means of the switch so that the voltages are series-aiding. Lateral-cut recordings result in the oscillatory angular motion of the coils which are also connected by the proper operation of the switch so that the voltages are again series-aiding. When the coils are connected by means of the switch for vertical reproduction, voltages generated by unwanted lateral modulation in the record are suppressed and when the coils are connected for lateral reproduction, voltages generated by unwanted vertical modulation are effectively suppressed.

The 5A reproducer arm was developed as a companion piece for the 109-type reproducer and permits the reproducer to be connected quickly, both mechanically and electrically, to the arm. Best results from the 109-type reproducer will be realized when used with the 5A reproducer arm inasmuch as they were designed for use with each other in order to provide optimum tracking conditions.

The balance of the arm allows a stylus pressure of 35 grams, contributing to long record life. The weight of the arm is approximately 3½ pounds and offers a sufficient mass to stabilize tracking over the useful frequency range.

The 5A reproducer arm is equipped with a four-prong type jack for engaging the four-pin type plug of the reproducer and a thumb screw on the reproducer is used for locking the reproducer in place in the arm jack. One end of four connecting wires is soldered to the terminals of the arm jack and the other end of the connecting wires, which project from the arm, are provided with spade-type cord tips for making connections to the terminals of the equalizing equipment.

The over-all length of the arm reproducer is 18½ inches and the distance between the stylus and pivot is 13⅛ inches. The height of the arm is adjustable to accommodate different heights of turntable platters.

KS 13386 Equalizer and 171 Repeating Coil. The KS 13386 equalizer and the 171A repeating coil form the equalizing, switching, and impedance-matching portion of the 109-type reproducing groups.

The equalizer switch has seven reproducing positions, two for vertical and five for lateral. The seven reproducing characteristics are based on two fundamental frequency-response characteristics, curve *A* and curve *B*, Fig. 5-10. These curves are based upon a survey of the recording field and match those which are currently used for record production.

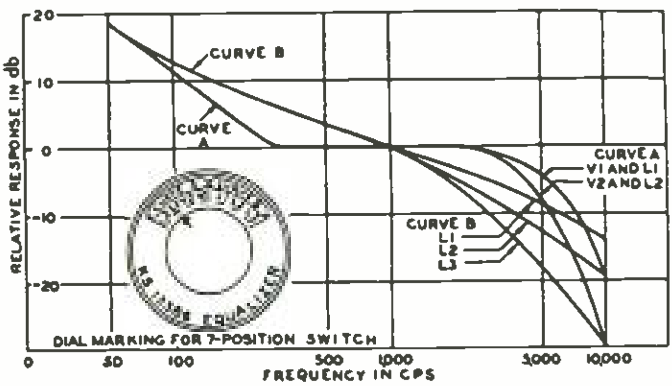


Fig. 5-10. The equalizer switch positions and frequency response curves corresponding to each position for the Western Electric KS13386 equalizer.

Courtesy Western Electric

Curve *A* is the conjugate of the frequency-response curve in general use in recording vertical transcriptions, some early lateral transcriptions and some "instantaneous-type" lateral transcriptions.

Curve *B* is the conjugate of the frequency-response curve in general use in recording lateral transcriptions (NAB Standard and Orthacoustic) and phonograph records. For information on specific curves used in making recordings, recording studios should be consulted.

The switch positions having the characteristic according to curve *A* should be chosen for the reproduction of records made with the characteristic which is the conjugate of this curve. Similarly, the switch positions indicated for curve *B* should be chosen for the reproduction of records made with the conjugate *B*. On Table 5-I following is indicated the frequency response which will be obtained for the various switch positions.

TABLE 5-I

	Equalizer Switch Position	Recording Type	Over-all System Response
Curve *A*	*V1*	Vertical	Uniform 50 to 10,000 cps
	V2	Vertical	Uniform 50 to 2,500 cps with roll off to 15 db down at 10,000 cps
	L1	Lateral	Uniform 50 to 10,000 cps
	L2	Lateral	Uniform 50 to 2,500 cps with roll off to 15 db down at 10,000 cps
Curve *B*	*L1*	Lateral	Uniform 50 to 10,000 cps
	L2	Lateral	Uniform 50 to 1,000 cps with roll off to 5 db down at 10,000 cps
	L3	Lateral	Uniform 50 to 1,000 cps with roll off to 17 db down at 10,000 cps

Curve *A*, *V2* and *L2*, and curve *B*, *L2* and *L3*, have a falling characteristic at the high-frequency end which is useful for the reproduction of records which have a high inherent surface noise, or have been worn so that surface noise becomes objectionable. In some cases, particularly in the case of sound effect records, it may be expedient to lower the response of high frequencies for a desired dramatic effect. Otherwise, curve *A*, *V1* and *L1*, and curve *B*, *L1*, are generally preferred.

The 171A repeating coil provides taps for feeding circuit impedance of 30/50 ohms, 150/250 ohms, or 500/600 ohms. The coil must feed directly into a resistive circuit such as a constant-impedance type mixer. If a constant-impedance mixer is not used, it is recommended that a simple "L" pad be inserted between the repeating coil and the input terminals of a preamplifier (unless the amplifier input presents a pure resistance to match the coil impedances). Schematic drawings of suggested L pads are shown in Fig. 5-11.

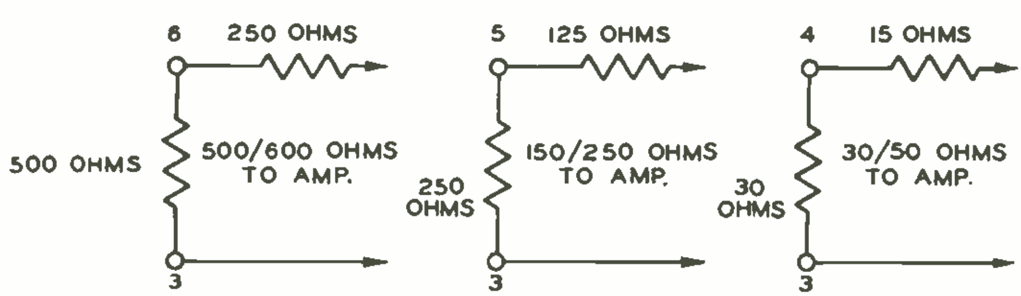
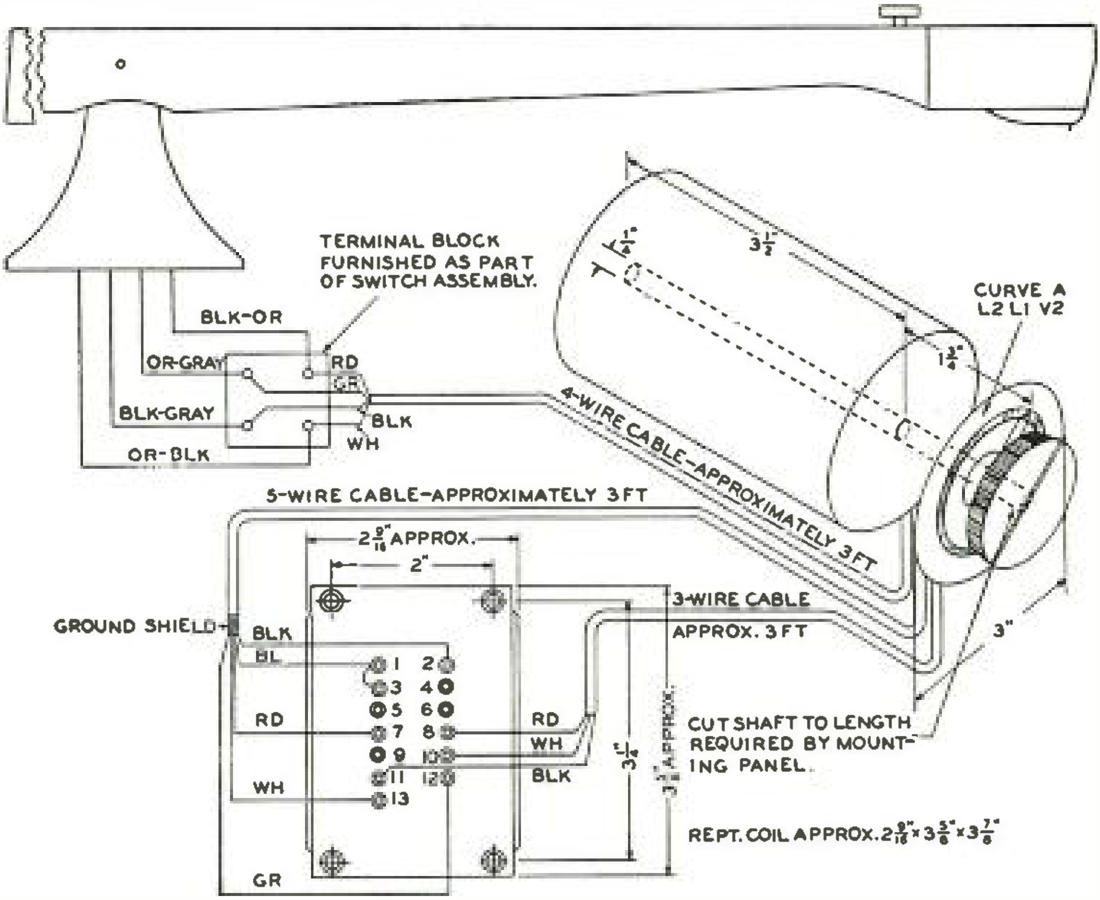


Fig 5-11. Schematic diagrams of L pads for use in matching the output impedance of the 171A repeating coil to the input of a preamplifier.

The 171A repeating coil should be located outside the range of alternating-current magnetic fields which may be produced by the turntable drive motor or amplifier power transformers. Mounting dimensions, color coding, and terminal designations are shown on Fig. 5-12.



Courtesy Western Electric

Fig. 5-12. The Western Electric 171A repeating coil and terminal board showing color code for connections.

The input circuits with which this reproducer group is connected should be operated unbalanced with respect to ground and should be connected to the 171A repeating coil as follows:

1. For 30/50 ohms — use terminals *3* (ground side) and *4*.
2. For 150/250 ohms — use terminals *3* (ground side) and *5*.
3. For 500/600 ohms — use terminals *3* (ground side) and *6*.

The 171A repeating coil must be terminated in a resistance. Amplifier input circuits presenting a reactive load will alter the frequency-response characteristics of the equalizer, so that it is necessary to use either a constant impedance mixer circuit or a fixed resistive pad between the 171A repeating coil and the amplifier to stabilize impedance conditions.

In all cases the ground should be connected to terminal *3*.

REPRODUCER GROUP WIRING CONNECTIONS

4-Wire Cable from Switch	109-Type Reproducer
RD	BLK-OR
GR	OR-GRAY
BLK	BLK-GRAY
WH	OR-BLK

3-Wire Cable from Switch	171A Repeating Coil
RD	Terminal *8*
WH	Terminal *10*
BLK	Terminal *11*

5-Wire Cable from Switch	171A Repeating Coil
BLK	Terminal *2*
BL	Terminal *1* and *3*
RD	Terminal *7*
GR	Terminal *12*
WH	Terminal *13*

The component parts of the 109-type reproducer are necessarily very light in weight and should be handled with care usually taken with any comparable delicate apparatus. Under normal use the reproducer will have a long useful life, but if it is accidentally dropped on the record, or allowed to skip across the record, the stylus point may be damaged. In this event, the reproducer should be returned for repair to the dealer from whom it was purchased. Occasionally dirt and dust from the record grooves will accumulate on the stylus, and this should be removed before it can interfere with tracking.

Contacts on the KS 13386 equalizer switch should occasionally be cleaned with a cloth moistened with a small amount of unmedicated vaseline. A slight film should be left to provide lubrication.

Precautions should be taken to maintain good electrical contact between the thumb screw of the 109-type reproducer and the 5A reproducer arm. If paint or dirt breaks the electrical connection, the case of the 109-type reproducer is not grounded and serious noise conditions may result.

Adjustment of the RCA pickup arm is described in the following instructions for the RCA 70-C2 turntable.

BROADCAST TURNTABLES

RCA 70-C2 Turntable[2]

The type 70-C2 transcription turntable utilizes a combination pickup and arm assembly for use in reproducing both lateral- and vertical-cut records. A removable door on the front of the cabinet permits easy access to the motor, filter, and terminal boards. There is sufficient space within the cabinet to mount a booster amplifier.

Motor and Turntable. The motor is a high-torque synchronous type, cushion-mounted on the bottom shelf of the equipment. The turntable has associated with it a separate flywheel to insure excellent speed regulation.

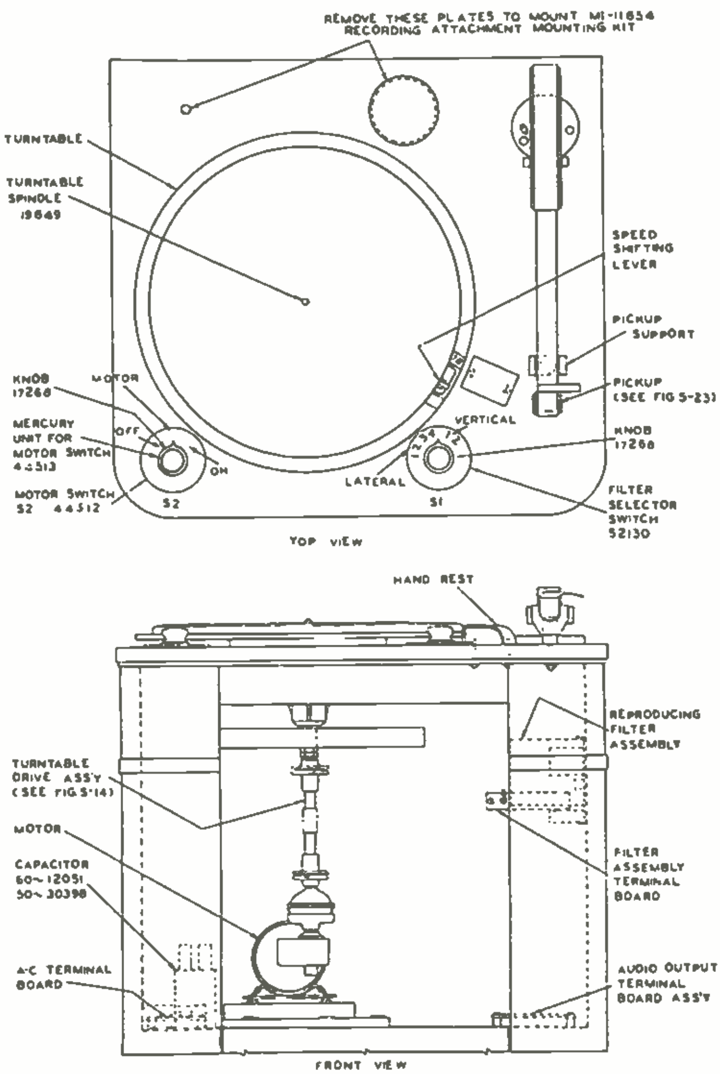


Fig. 5-13. Top and front views of the RCA 70-C2 transcription turntable for reproducing both lateral- and vertical-cut records. A high-torque synchronous motor provides 78 and 33⅓ rpm turntable speeds. The pickup head is of the moving-conductor type with a diamond point stylus. The pickup and filter reproduce various types of records with a response characteristic considered an ideal playback response.

Courtesy RCA

[2] Courtesy RCA.

Power Switch. The a-c power switch is a mercury tube of the tumbler type. This silent type of power switch, mounted atop the cabinet, permits the turntable to be operated near a microphone.

Turntable Speeds. A speed-change switch near the edge of the turntable plate is used to select either 78 or 33⅓ rpm for the turntable speed. See Fig. 5-13.

Pickup and Filter. The pickup and filter reproduce the various types of records (Orthacoustic, Victor, RCA, Columbia, World, and others) with a response characteristic which is considered an ideal play-back response. The filter unit is designed so that this may be accomplished by turning a switch to one of six positions.

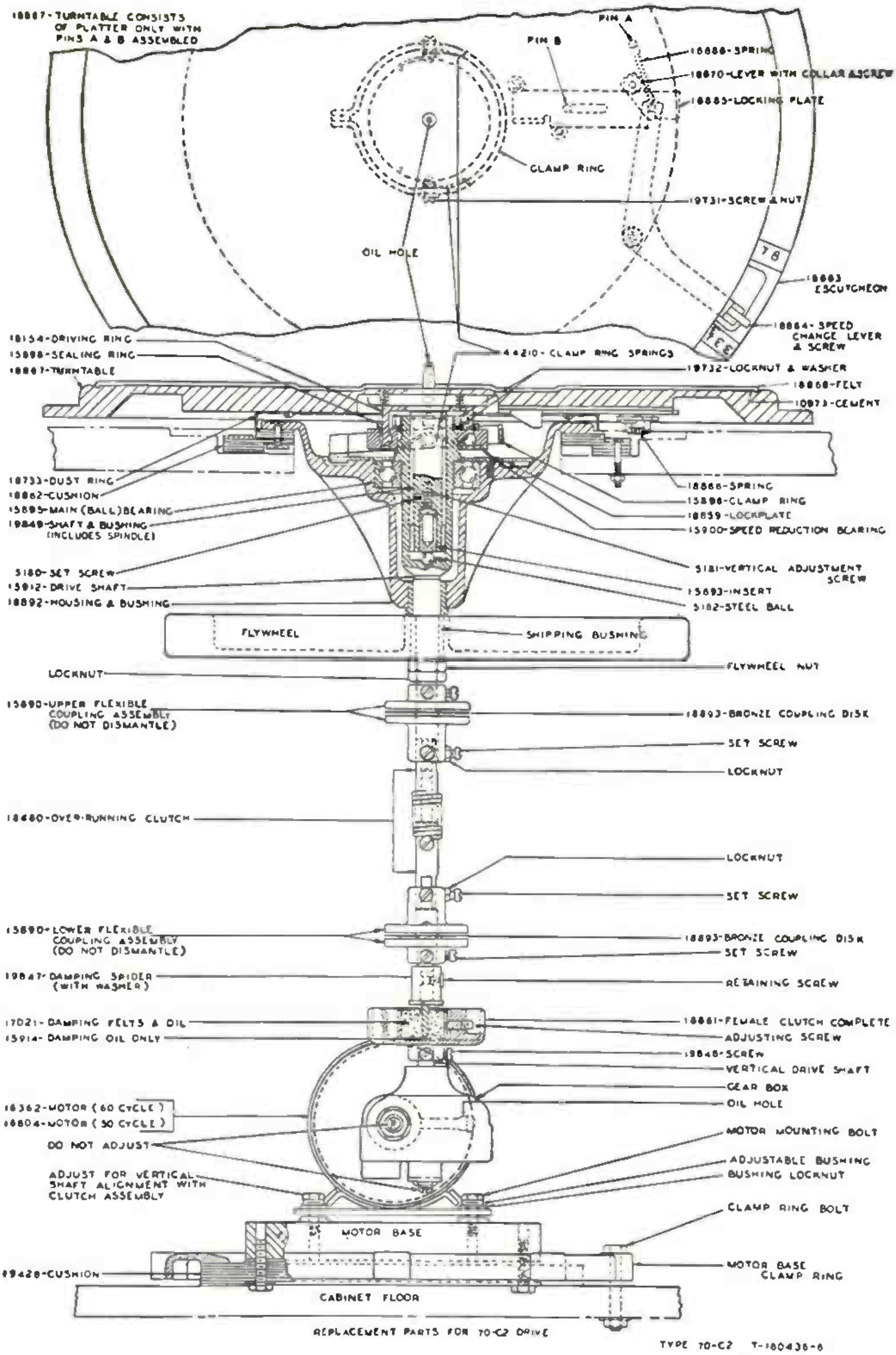
The pickup head is of the moving-conductor type in which two ribbons are free to vibrate in their respective magnetic fields. The phase relation of the voltage generated in the ribbons reverses 180° when changing from reproducing lateral records to vertical records. The filter unit in the output circuit selects the proper circuit so that the output voltages will be additive for either type of record.

Since any vertical modulation will be cancelled out when playing lateral-cut records, the "pinch-effect" in lateral reproducing is cancelled out. (The pinch-effect is due to cutting a record with a plane and reproducing with a spherical surface. This produces second-harmonic frequencies which would seriously affect the quality of reproduction if not cancelled out by the pickup filter circuit.)

The pickup head contains a diamond-point stylus to eliminate the necessity of changing styli.

Fig. 5-14 shows the turntable drive assembly. The motor and the entire drive-shaft assembly are carefully aligned at the factory. Under no circumstances attempt to realign the assembly by loosening the motor mounting bolts in the motor mounting ring. The horizontal and the vertical shaft thrust bearing adjustment screws in the motor gear box are likewise carefully adjusted at the factory, and under no circumstances should they be disturbed.

The mechanical filter (cup-like receptacle above the motor) includes an adjusting screw which, when tightened, causes movable plates to compress the felt cushions which form a part of this filter. As shipped from the factory this adjusting screw is loosened an amount such that the proper pressure is obtained when the screw (see Fig. 5-14) is tightened six turns. At this point, the felt cushions should be firm but not hard.



Courtesy RCA

Fig. 5-14. The turntable drive assembly of the RCA 70-C2 transcription turntable. The motor and drive shaft assembly are carefully aligned at the factory.

Lubrication. Pour the entire contents of the bottle of oil shipped with the equipment into the oil well, which is the cup-like receptacle of the mechanical filter. The purpose of this oil is to saturate the felt cushions accommodating the two driving vanes and thus lubricate the assembly. Under normal conditions it is not necessary to replace this oil. If, however, circumstances arise making such replacement necessary, use a similar quantity of Vacme "AA" oil or its equivalent (SAE-60). Two oil pipes are provided, one in each end-bell, for the lubrication of the motor bearings, and an oil hole is provided in the

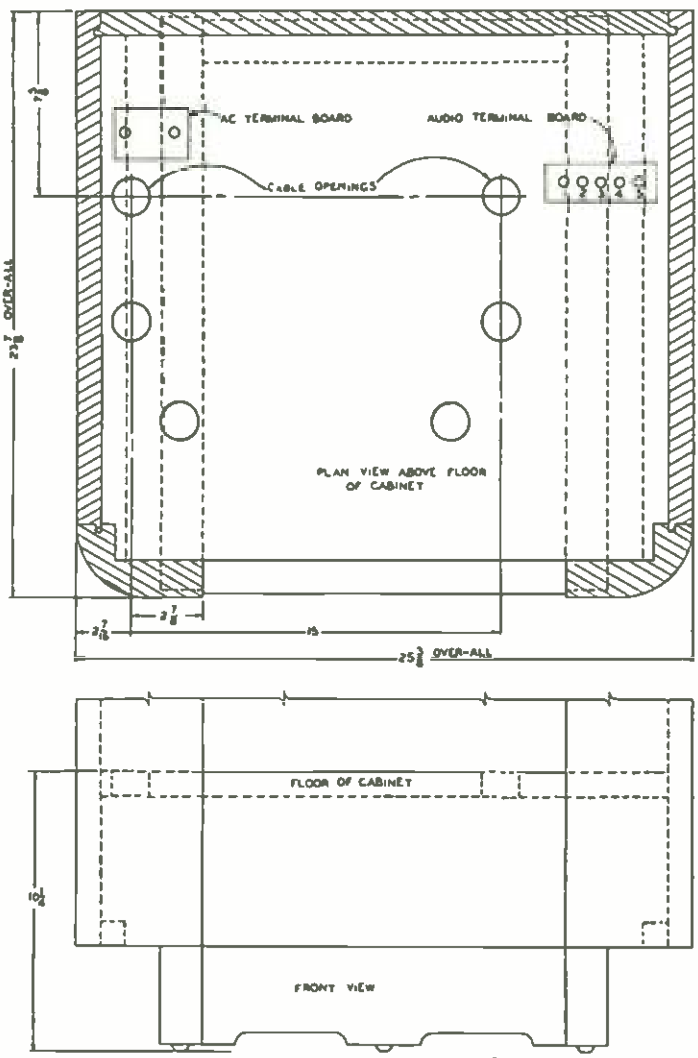
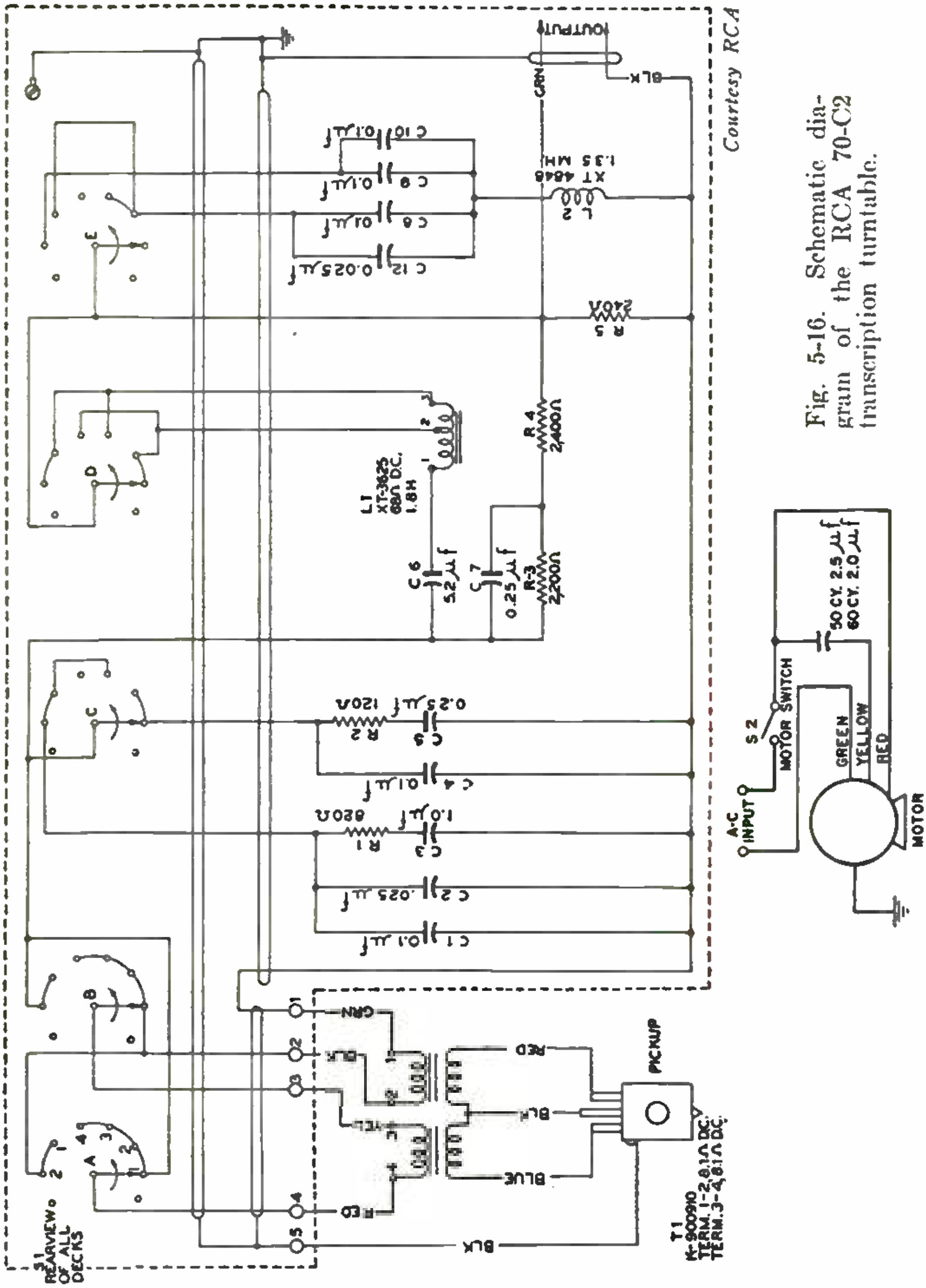


Fig. 5-15. Positions of a-c and audio terminal boards and location of cable openings in the RCA 70-C_2 transcription turntable. The voltage and frequency of the a-c power supply are specified on the nameplate. Use a shielded twisted-pair cable for audio output connections.

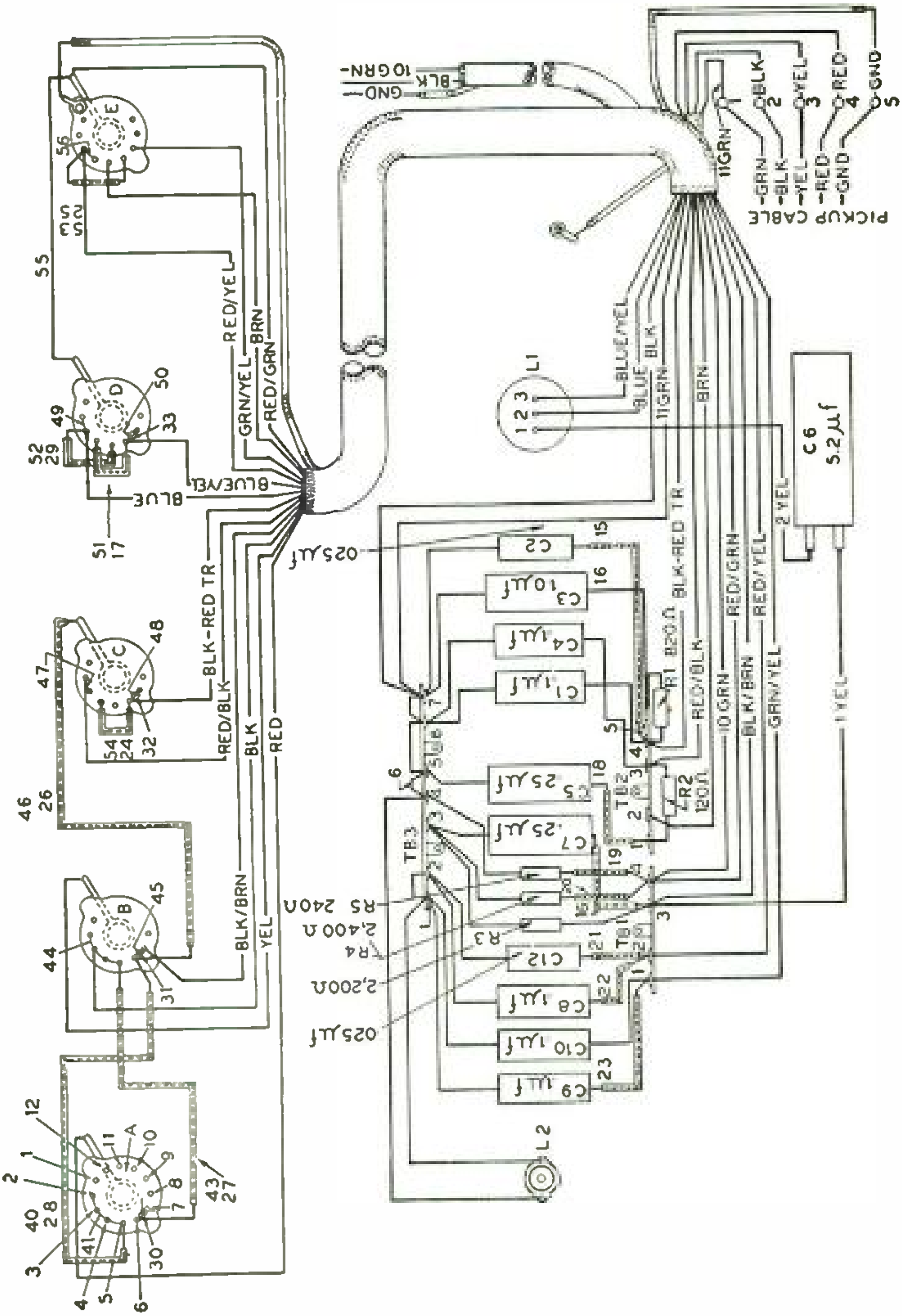
Courtesy RCA

top turntable spindle for lubrication of the spindle bushings. Use a light (SAE-20) high-grade nongumming machine oil, and apply six or seven drops to the motor bearings and a few drops to the turntable spindle oil home at intervals of one month. Before oiling the turn-



Courtesy RCA

Fig. 5-16. Schematic diagram of the RCA 70-C2 transcription turntable.



Courtesy RCA

Fig. 5-17. Wiring diagram for the RCA 70-C2 transcription turntable.

table, remove the pin stopper, and after oiling replace the stopper to prevent dust and dirt from clogging this hole.

Location. Be sure the turntable is installed on a level surface. A three-point support is provided to eliminate the possibility of the turntable rocking when the floor is uneven.

Recommended Associated Equipment. The pickup filter is designed to work directly into the primary or an unloaded input transformer, and may be used with any amplifier having a low-impedance input, unloaded input transformer, and flat frequency-response characteristic.

Connections of RCA 70-C2

Caution. Be sure that the motor switch is OFF when making connections.

Refer to Fig. 5-15 for the position of the a-c and audio terminal boards and for the location of cable openings. Make the required connections to an a-c power supply of the voltage and frequency specified on the nameplate. If rigid conduit is used, be sure no part of the conduit touches the cabinet.

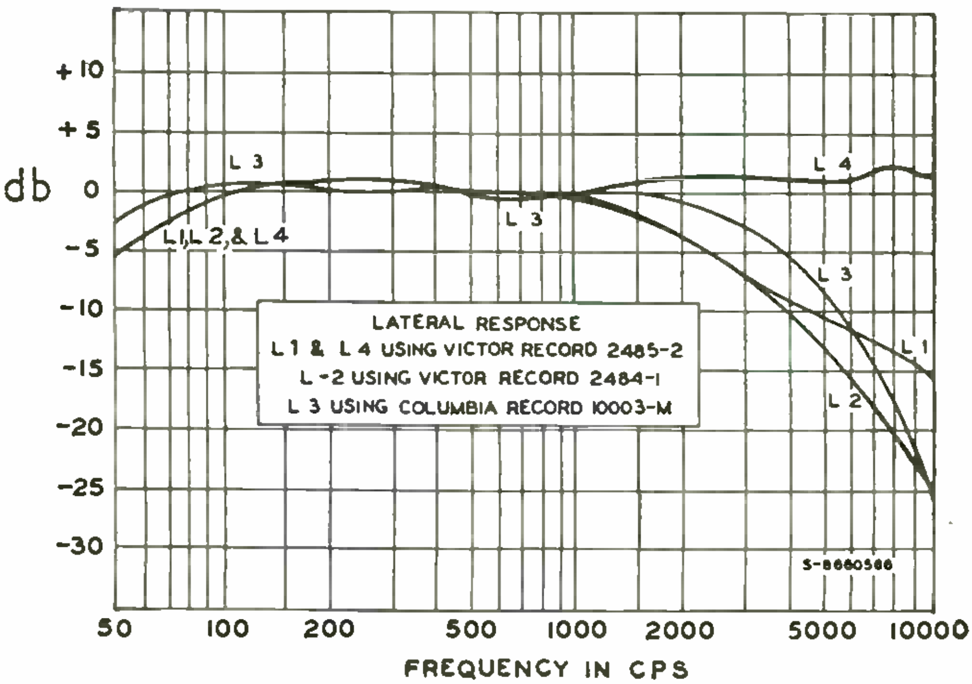
The audio-output terminal board contains five terminals. Terminals *1* and *2* are the high and low sides, respectively, of the audio output. Terminal *3* is the ground terminal and should be connected to a good ground such as the station ground, a cold water pipe, etc. Terminals *4* and *5* are not connected at the factory, but may be used for the connection of other filters and accessories. Use a shielded twisted pair of No. 19 AWG for the output connections. Ground the cable shield.

Two unmarked positions on the pickup filter switch may be made available for connecting additional filters by moving the switch stops. Schematic and wiring diagrams are shown in Figs. 5-16 and 5-17.

Operation of the RCA 70-C2

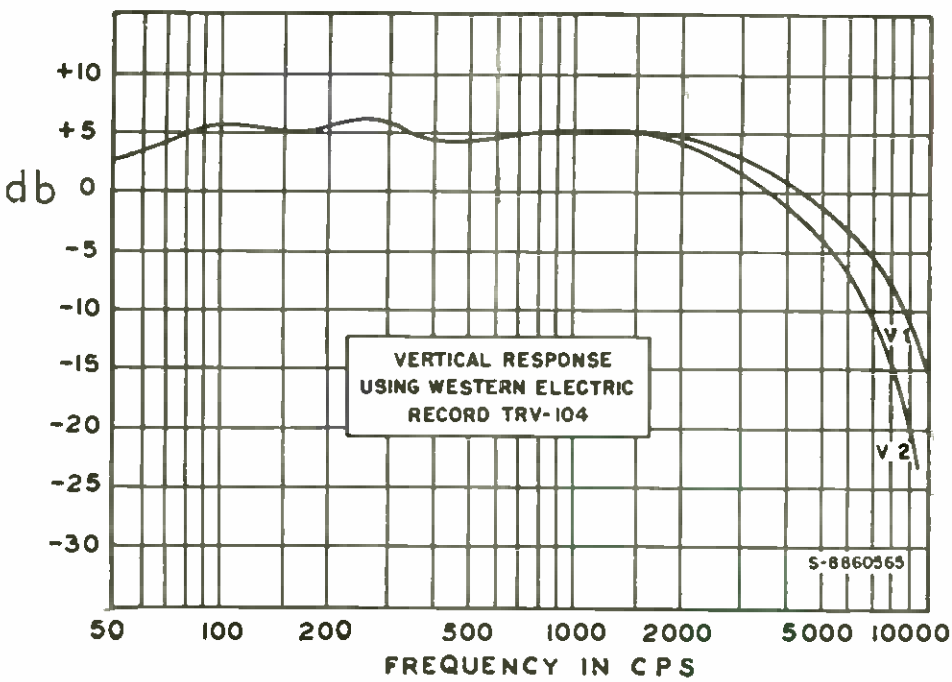
Frequency Response. The filter switch dial has six positions, four for lateral reproducing and two for vertical reproducing. Refer to Figs. 5-18 through 5-21 for the various response curves using record and oscillator inputs. Fig. 5-22 illustrates the connections to be made to secure the oscillator curves.

Note. The absolute level corresponding to 0-db response depends on a number of factors, an important one being the level employed in recording. However, it is probable that a good average is about the



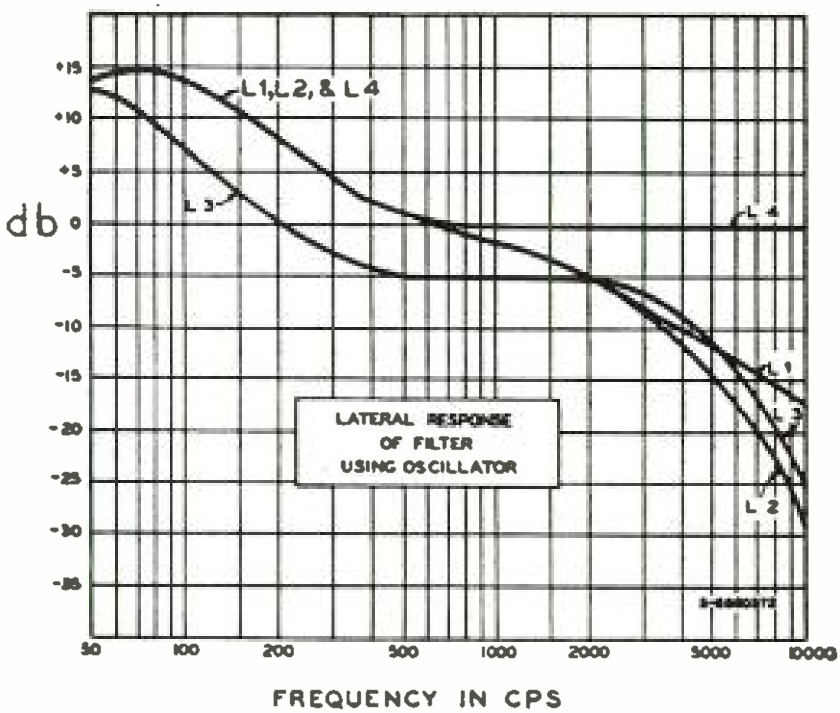
Courtesy RCA

Fig. 5-18. Frequency response for lateral switch positions using test record.



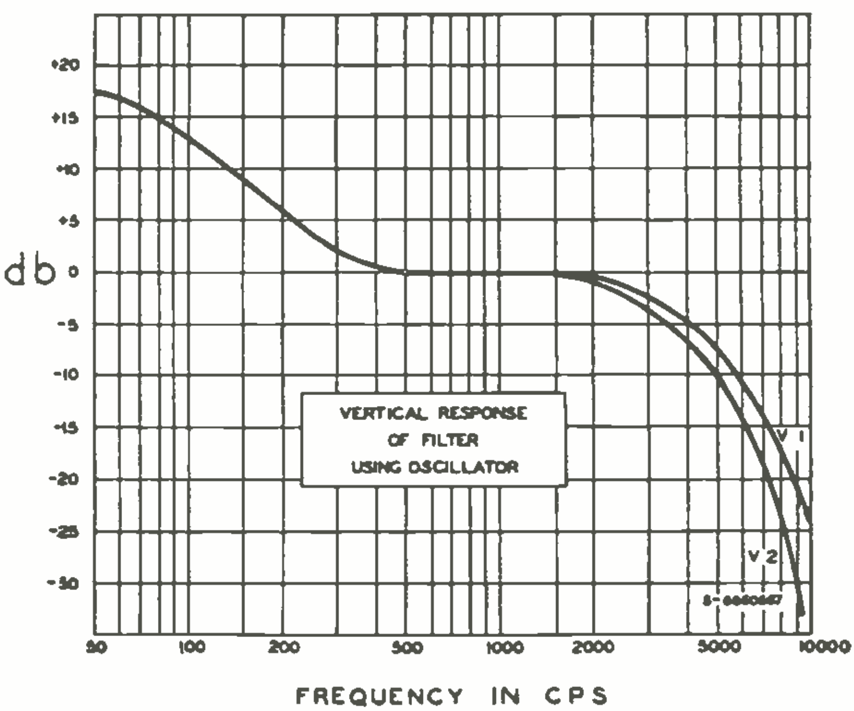
Courtesy RCA

Fig. 5-19. Frequency response for vertical switch positions using test record.



Courtesy RCA

Fig. 5-20. Frequency response for lateral switch positions using oscillator.



Courtesy RCA

Fig. 5-21. Frequency response for vertical switch positions using oscillator.

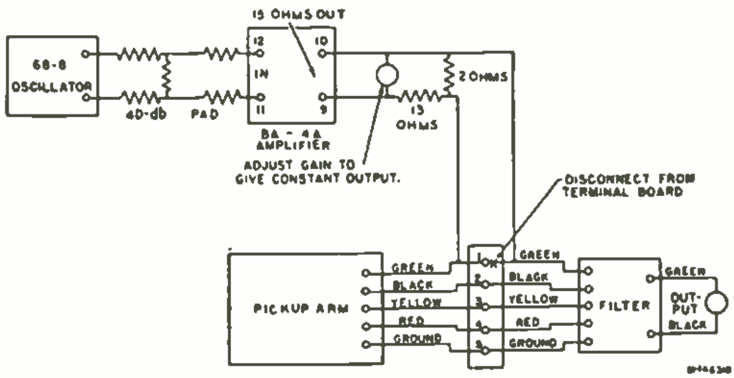


Fig. 5-22. Connections of circuit used to obtain frequency response curves with oscillator input.

Courtesy RCA

same as that accepted for the dynamic microphone on program pickup, namely, −62 db below 1 mw approximately.

Switch Positions. The switch positions are to be used as follows:

Switch Position *L1:* Lateral transcriptions, for reproduction of transcriptions recorded according to the NAB standard lateral characteristic.

Switch Position *L2:* Phonograph records, for reproduction of records and transcriptions which have been recorded with a cross-over frequency of 500 cps, such as Victor.

Switch Position *L3:* Phonograph records, for reproduction of records and transcriptions which have been recorded with a cross-over frequency of 300 cps.

Switch Position *L4:* Flat response, for reproduction using the automatic equalizer to raise the high-frequency level at the inside of the record. This position may also be used for test purposes.

Switch Position *V1:* Vertical transcriptions, such as World for reproduction of transcription recordings cut in accordance with the NAB standard vertical characteristic.

Switch Position *V2:* Vertical transcriptions such as Associated, same characteristic as *V1* except less high-frequency response.

In order to insure stability of operation, turn the motor switch ON and allow turntable to run for at least five minutes before playing a record. This should be done especially when the instrument has been idle for an appreciable period (such as overnight or when the instrument has been exposed to the cold), but it is not essential between operations separated by intervals of short duration.

After setting the speed-shifting lever, rotate the turntable platter slowly by hand until the mechanism is heard to engage. Avoid engaging the mechanism by holding the turntable platter and starting the motor.

Cuing: For close cuing of records, it is satisfactory to hold the record by finger pressure on the rotating platter and to release the record just prior to opening the fader for the turntable channel. This practice may shorten the life of the turntable felt, but a new turntable felt may be ordered which is installed easily.

Another common cuing practice is to locate the pickup in a starting groove and rotate the record until the program is heard to start either by direct "talkback" from the record or through an electrical cue system. Without lifting the pickup stylus from the record, move the disk back by a portion of a revolution to permit the turntable to come up to speed before the program starts after the motor switch is turned on. This may be over ¼ revolution for 78 rpm records and over ½ revolution for 33⅓ rpm recordings.

Maintenance of the RCA 70-C2

Lubrication. Lubricate the equipment in accordance with the instructions given earlier. Do not oil the over-running clutch. This clutch is lubricated at the factory.

Cleaning. Do not blow dust from beneath the record plate. (The speed-reduction bearing is protected from the normal accumulation of the dust from above, but is liable to contamination if dust is blown upward from the recess in the cabinet in which the bearing is mounted.) Wipe the dust from beneath the record plate with a lint-free and slightly oily cotton cloth.

Parallel Adjustment of Pickup Arm. The pickup arm, Fig. 5-23, should be parallel to the record face and is so adjusted at the factory. Should this at any time not be the case, proceed as follows: Determine the amount of deviation from the parallel. Then disconnect the pickup leads from the terminal board and remove the pickup cable from the clamp above the terminal board. Remove the pickup arm assembly from the cabinet top. Remove the two machine screws (*A*, Fig. 5-23). Pull out the two pins and remove the spacers at the pivot. Lift the pickup arm from the base casting. This exposes the upper fulcrum screw (Fig. 5-23). Loosen the locknuts on both upper and lower fulcrum screws and adjust the screws to raise or lower the pivot casting as required. A complete turn of the screw produces a vertical motion of $\frac{1}{32}$ inch. Retighten the locknuts so that the pivot casting does not bind and there is no side or end play in the bearings. Install the arm and cable in their former positions.

Horizontal Adjustment. A horizontal stop screw and locknut are mounted on the side of the base casting (Fig. 5-23). This screw has been adjusted at the factory to prevent the needle from reaching the inner diameter on which the record driving holes are placed. This is to prevent possible damage to the stylus if it were to drop into these driving holes. If necessary, adjust the screw and nut to prevent excessive horizontal movement of the arm.

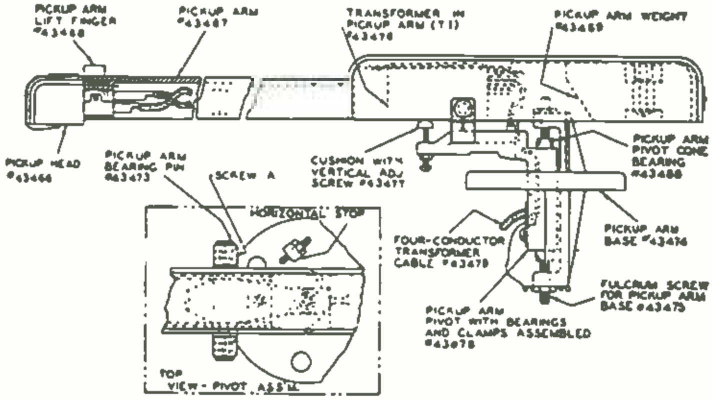


Fig. 5-23. Side view of the pickup arm assembly. The insert shows the top view of the pivot assembly.

Courtesy RCA

Weight Adjustment. The pickup arm is adjusted at the factory for a weight on the record of 28 (± ½ grams) at the needle point. The high-frequency response of the pickup is somewhat dependent on this weight. If at any time it is found that the weight is otherwise proceed as follows: Loosen the screw which holds the lead counterweight in place under the rear of the arm. To increase the weight, move the counterweight forward. To decrease the weight, move the counterweight toward the rear. Retighten the screw.

Vertical Adjustment. A vertical adjusting screw and locknut with a rubber cushion or stop on the end of the screw is placed in the pivot casting. (Fig. 5-23.) The height of this cushion is adjusted at the factory so as to allow the needle just to graze the felt at the top of the turntable. If necessary, adjust this screw to obtain the correct pickup arm height.

Care of Turntable. As a result of the continued demand for higher and higher quality in the reproduction of broadcast transcriptions, the transcription turntable has been developed, through engineering refinements, into a device in the class with precision instruments and should be treated as such. With reasonable care and proper lubrication, as outlined below, it should give years of constant service without noticeable increase in speed variation. Therefore, do not tamper with, or alter in any way, any part of the turntable drive assembly or record plate, or change any adjustments of this mechanism, unless it is absolutely necessary because of a known defect.

Clutch Adjustment of Speed Changing Mechanism.

Note. Do not remove the turntable unless this adjustment (or that as described below) is necessary.

The distance from the top of the turntable to the top of the cabinet should be $1\frac{5}{32}$ inches. If this distance should ever vary, scraping and speed variation might result. If at any time it becomes necessary to reset this distance, the turntable may be removed from its bearing and readjustment of the clutch mechanism may be made in the following manner:

1. Grasp the turntable with the hands at diametrically opposite points on its circumference and withdraw it from its bearing by exerting a straight, upward pull. When doing this be sure to hold the turntable in a level position until its spindle is entirely clear of the bearing. Otherwise, damage of the bearing may result.
2. The steel ball which serves as a thrust bearing under the end of the turntable spindle may adhere to the grease on the spindle and be removed from its seat. Be careful not to lose this ball.
3. Place the turntable, face down, on a clean level surface.
4. Remove the cylindrical plug in the end of the turntable spindle.
5. A vertical bearing adjustment, slotted to accommodate a screwdriver, will be found at the bottom of the hole from which the plug has been removed. A set screw in the side of the shaft serves to clamp this adjustment.
6. Loosen this set screw, back out or screw in the adjustment, and insert the plug.
7. See that the steel ball thrust is in its seat in the turntable spindle bearing in accordance with instructions given below, and test for correct adjustment, the conditions for which are given in subparagraph 8. below.
8. The clutch engagement should be accomplished in the first revolution of the turntable, and the equipment must not be operated if the clutch slips.

Cleaning the Speed Reduction Bearing. If the equipment is not kept clean or is cleaned improperly, dust may become lodged in the speed reduction bearing. If the dust is permitted to remain and accumulate, excessive wear and possible speed variation will result. Dust in the speed reduction bearing will be indicated by a faint knock-ing or grinding noise, heard directly from the mechanism when running at 33⅓ rpm. This dust will cause a rumble in the output of the system. This rumble will be detected most easily by playing a silent record and listening to the monitoring speaker. To eliminate this rumble, it is necessary to remove the bearing and clean it thoroughly.

Do not raise the turntable or remove the bearing unless it is necessary to perform this work (or that described above). The procedure is as follows:

1. Remove the turntable as described above.
2. Unlock the bearing nut by bending the lockwasher tabs outward and away from the notches in the bearing nut.
3. Remove the bearing nut, being careful not to damage the bearing. A spanner wrench is recommended for this purpose.
4. Remove the lockwasher and sealing rings.
5. Lift the speed reduction bearing clear of the bearing housing without removing the clamp ring.
6. Clean all parts on the top of the housing of the mechanism and remove all dust from the top of the cabinet.
7. Soak the speed reduction bearing thoroughly in clean kerosene oil, turning the steel balls and ball races to be sure that no dirt or lint remains on the assembly.
8. Wash the bearing in a second rinse of clean kerosene and shake off all kerosene possible. Do not attempt to dry the bearing with a cloth, as this operation may reintroduce lint to the bearing parts and cause a repetition of the trouble.
9. Apply a thin coating of pure, clean, white petroleum jelly to the ball race. Petroleum jelly sold in a tube type container is suggested, since there is almost no possibility of its becoming contaminated or dirty.
10. Replace the bearing on the bearing shaft. See that the notch in the clamp ring is uppermost when the assembly is in position.
11. Place the lockwasher on the bearing shaft.
12. Place over the lockwasher a clean piece of paper, having in its center a round hole the size of the bearing shaft, and large enough to protect the ball race.
13. Screw the nut in place and tighten it with a spanner wrench.
14. Lock the nut by bending a lockwasher tab into a notch in the bearing nut.

15. Remove the paper mentioned in 12. above.

16. Place the sealing ring in position on top of the speed reduction bearing and center it.

17. Clean the underside of the turntable and replace the spindle in the bearing as described below.

Replacement of Turntable. If, for any reason, the turntable has been removed from its bearing in the main drive spindle, replace it in the following manner:

1. Engage the turntable spindle with the female bearing of the drive, align the spindle, and lower the turntable slowly until the upper member of the ball race, which is attached to the turntable, comes into contact with the balls. When performing this operation make certain that the dust ring, which rests on the top of the spindle, is concentric with the bearing.

2. While still holding the turntable, rotate it slowly until the openings in the upper member of the ball race engage the balls, and then lower it slowly and carefully into position. Be sure to avoid causing flats on the balls.

Speed Variation. A variation in the speed of rotation of the turntable, sometimes referred to as "wows," can be caused by any of the following:

1. Grease on the over-running clutch. This may be removed by any usual cleaning means, such as cleaning thoroughly with carbon tetrachloride.

2. A loose clamp on the speed-reduction bearing. Loosen the clamp nuts and tighten the two screws which hold the clamp in place. Retighten the nuts.

3. Worn gears in the motor gear box. This is corrected by replacing the motor as described below.

4. Improper alignment of motor shaft and spindle. In general, the optimum position is that which gives minimum undulation of the bronze coupling disks in the flexible coupling assemblies. See below for the method of adjustment.

5. Loose flywheel. Be sure the flywheel nut and locknut are well tightened.

Changing the Motor. To replace the drive motor, disconnect electrically and proceed as follows:

1. Loosen the lock screws on the upper and lower flexible coupling units and on the spider of the mechanical filter.

2. Remove the four motor mounting bolts.

3. Raise the entire spider and coupling assembly to clear the vertical drive shaft and set the upper flexible coupling lock screws and spider lock screws so as to hold the assembly clear of the vertical drive shaft.

4. Carefully lift out the motor.

5. Remove the mechanical filter from the vertical drive shaft and place it in the same position on the vertical drive shaft of the new motor.

Note. A washer in the mechanical filter is shown in Fig. 5-14, turntable drive assembly. This washer can easily be forgotten or lost before the reassembly is started. Therefore, be sure to locate this washer and set it safely aside at the time of removing the old motor.

6. Place the new motor carefully on the motor base and lock the mechanical filter to the vertical drive shaft. Flats are provided on the shafts for the lock screws.

7. Align the vertical shaft with the coupling shaft and start, but do not tighten, the motor mounting bolts.

8. Loosen the upper flexible coupling and spider lock screws and lower the coupling unit assembly into place so that the spider engages the female section of the mechanical filter.

9. Lock the lower and upper flexible coupling lock screws to the flats of the shafts to which they fit.

10. Make electrical connections to the new motor.

11. Oil the new motor as described previously.

12. Start the motor and observe the bronze coupling spring in the lower flexible coupling unit. The spring will run horizontally between the coupling unit sections and will appear as a plane surface when the motor is aligned properly. If the motor is not correctly in line, the bronze spring will show a definite undulation as it turns.

13. Adjustable motor base bushings with lock nuts are supplied on the motor. These bushings permit the raising or lowering of any corner of the motor and must be adjusted for the alignment of the vertical drive shaft and clutch assembly. The bushings should be locked after they have been set for alignment as described.

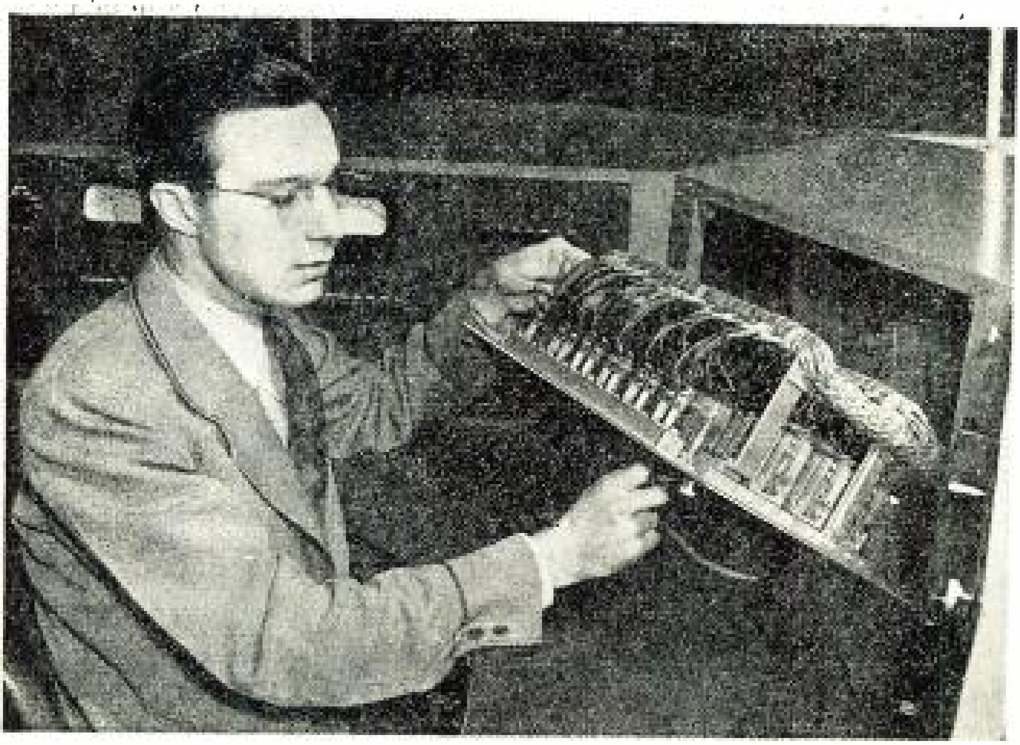
14. Tighten the motor mounting bolts while watching the bronze coupling spring with the motor running. Any variation in the bronze coupling spring movement will indicate that further adjustment of the motor base bushing will be necessary.

15. When the adjustable bushings are set so that the tightening of the motor mounting bolts does not throw the vertical drive shaft and clutch assembly shaft out of line, they are correctly set.

Note. The over-running clutch must be absolutely free to turn in the clockwise direction and must not bind. It must lock when turned counterclockwise.

Jack Panel Maintenance

Jacks, since they constitute either series or parallel connections in the signal path, must be kept free of dust and dirt, and in perfect contact adjustment. They should be vacuum cleaned frequently, or better still, a thin high-pressure stream of forced air should be used through each sleeve from front and rear, followed by a thorough vacuuming. Jack contacts may be kept clean by regular insertions and removals of patch-cord plugs as shown in Fig. 5-24. Visual inspection of jacks and mounting structures should be made during this process to check for proper operation and tightness of mounts. If jacks have *not* been maintained in this manner for a long period of time, a regular jack burnishing tool, such as manufactured by Western Electric, should be used. When the burnishing tool is used,



Courtesy WIRE

Fig. 5-24. Jacks must be kept free of dust and dirt. The jack contacts above are being cleaned by insertion and removal of patch-cord plugs.

the handle should be adjusted in such a way that about 2⅛ inches protrudes which prevents the blade from contacting the insulation in the spring pileup and avoids damage to the insulation. The blade should be inserted carefully in the jack until it barely meets the normal contact. A slight turning movement of the tool will then tend to lift the normal spring from the normal contact and a light forward pressure will cause the blade to slip easily between the contacts.

Key and Switch Maintenance

Keys and switches vary considerably in structure and functioning, but all types consist of a means of opening and closing single or multiple circuits by spring contacts and blade assemblies. Dirt or dust is the cause of the majority of troubles here. Contacts may usually be cleaned by using a clean, dry, lint-free cloth. A *clean* strip of silk is helpful since it will collect dust particles from the contacts by static electricity upon being drawn through the contacts with a light pressure being held by fingers on the blades or spring sheaves. In more serious conditions, a toothpick immersed in carbon tetrachloride should be drawn through the contacts then discarded to prevent reusing.

Mounting of keys or switches should be checked for tightness. Tighten every loose connection. Visually observe the mechanism when key or switch is operated, and note tension of spring or stationary contacts. If lost tension is apparent, adjust with fingers or long-nose pliers, or use the special switch and relay tool with a slotted end. Be very careful not to overdo any adjustment.

Maintenance of Faders (Attenuators)

A regular cleaning schedule is absolutely necessary for continued reliable and noise-free operation of fader controls. Contacts should be cleaned with a clean soft cloth and carbon tetrachloride. The leaves should also be cleaned in this manner, being careful not to disturb tension of the individual leaves. Some engineers recommend light lubrication of the contacts by applying a very slight amount of unmedicated vaseline from a clean cloth to help prevent wear.

Never disturb the tension of the leaves against the contacts unless absolutely necessary. If the attenuator must be dismantled for any reason, the leaves should be adjusted by the sliding mounting screw until just enough pressure is maintained to allow reliable contact.

Attenuators that turn "hard" against the fingers have excessive pressure beween leaves and contacts, and will cause shortened attenuator life.

Relays. See Chapter 17 on Preventive Maintenance Instructions for complete maintenance details on all types of relays.

Location of Amplifier Troubles

This text is not a detailed treatment of servicing techniques and a fundamental radio background is assumed from the beginning. However, for the purposes of completeness in studio maintenance, helpful hints are given here for locating troubles in audio amplifiers.

Filaments Do Not Light. When all filaments fail it may generally be assumed that either the a-c power line has failed, or a fuse has blown in the amplifier rack or on the amplifier itself. First check all a-c switches for proper operating position, then position of any circuit breakers involved, then fuses. Faulty switches may nearly always be detected by "feel" when operating them. If "snap" is not apparent on operation, jumper the contacts until switch can be replaced.

If only one tube fails to light, it is usually a bad tube. First "wiggle" the tube with strong pressure against the socket. If the filament comes on, a loose or "rosin connection" on the socket terminals may exist, or the contact springs may not have enough tension to create reliable contact with the tube prongs.

See Chapter 17 for detailed discussion of vacuum tubes and socket maintenance.

Filaments Light Up, No Response. First replace rectifier tube with one known to be good. If trouble still persists, plate voltages should be checked starting with the output of the filter system, then output stage, and so on back to first stages. If all voltages are normal, check continuity of output transformer. The best procedure from here on (assuming normal plate voltage) is to use a signal generator and headphones (with series 0.1 μf capacitor) to trace signal to faulty stage. If no plate voltage exists or voltage is very low, check condition of electrolytic capacitors and continuity of filter circuits. Check wiring for continuity. Check condition of power-supply switch and fuse holders. If voltage does not exist or is low in only one stage (in which case some distorted signal is usually present), check continuity and value of all resistors in plate and cathode circuit, and the wiring to the component parts. In some cases of low plate voltages, check for excessive plate currents. This may be caused by low or non-existent grid bias, or a leaky coupling capacitor placing a positive voltage on the grid of the following stage. Point-to-point analysis is called for, and any of the recognized methods of servicing such as signal tracing, point-to-point resistance measurements, etc. are required.

Amplifier Noise. Hum and noise are commonly encountered troubles. First, try to locate approximate origin of the noise from normal operating positions. For example, is the noise apparent with all switches and attenuators off? If so, the noise could be in the monitor amplifier or high-level amplifier circuits. It will, of course, vary according to the particular installation layout. Assume, however, that we are working with the Western Electric 23-C consolette as shown in the block diagram of Fig. 5-2. With all keys "OFF" or in neutral position (to cut off all microphone inputs) and all attenuators closed except the master fader, we get quiet operation. Now suppose that opening any or all of the attenuators (with mike keys still "off") brings in a noise or hum as apparent in our monitor speaker and (if of high enough level) by vu indicator meter. We know in this case that the noise is coming from the preamplifier stages since, had the noise been in either the low-level (with master fader open) or high-level amplifier, we would hear it at all times. If the noise were originating in the low-level stage, closing the master fader would kill it. If originating in the high-level stage (main program amplifier), we would not be able to kill the noise by any fader control. It is obvious here that we can gain an approximate location of originating noise or hum in any installation by following this procedure in relation to the specific type of circuit continuity.

It is always best to first replace tubes in case of noise. If the trouble does not clear by this procedure, look for loose connections or any apparent "rosin" connection. Move wiring back and forth with monitor amplifier turned up high. "Wiggle" tubes in their sockets. If faulty spring contact exists, see Chapter 17 for socket maintenance. Tap all coupling capacitors and resistors. If a distinct electrical noise (as distinguished from acoustic noise) exists, replace the part with an exact replacement. Examine closely all ground connections for looseness or bad soldering.

Steady noise that is not intermittent may be caused by some form of inductive interference. If input transformers are constructed in such a way that their windings may be oriented in any direction, try reorientation for minimum noise pickup from any possible a-c field. Some microphones are unusually sensitive to stray a-c fields. If noise is coming from any particular mike circuit, it is wise to check mike and cable location for possible interference. This is particularly important at remote-control locations.

Part 2

OPERATING THE MASTER CONTROL

Chapter 6

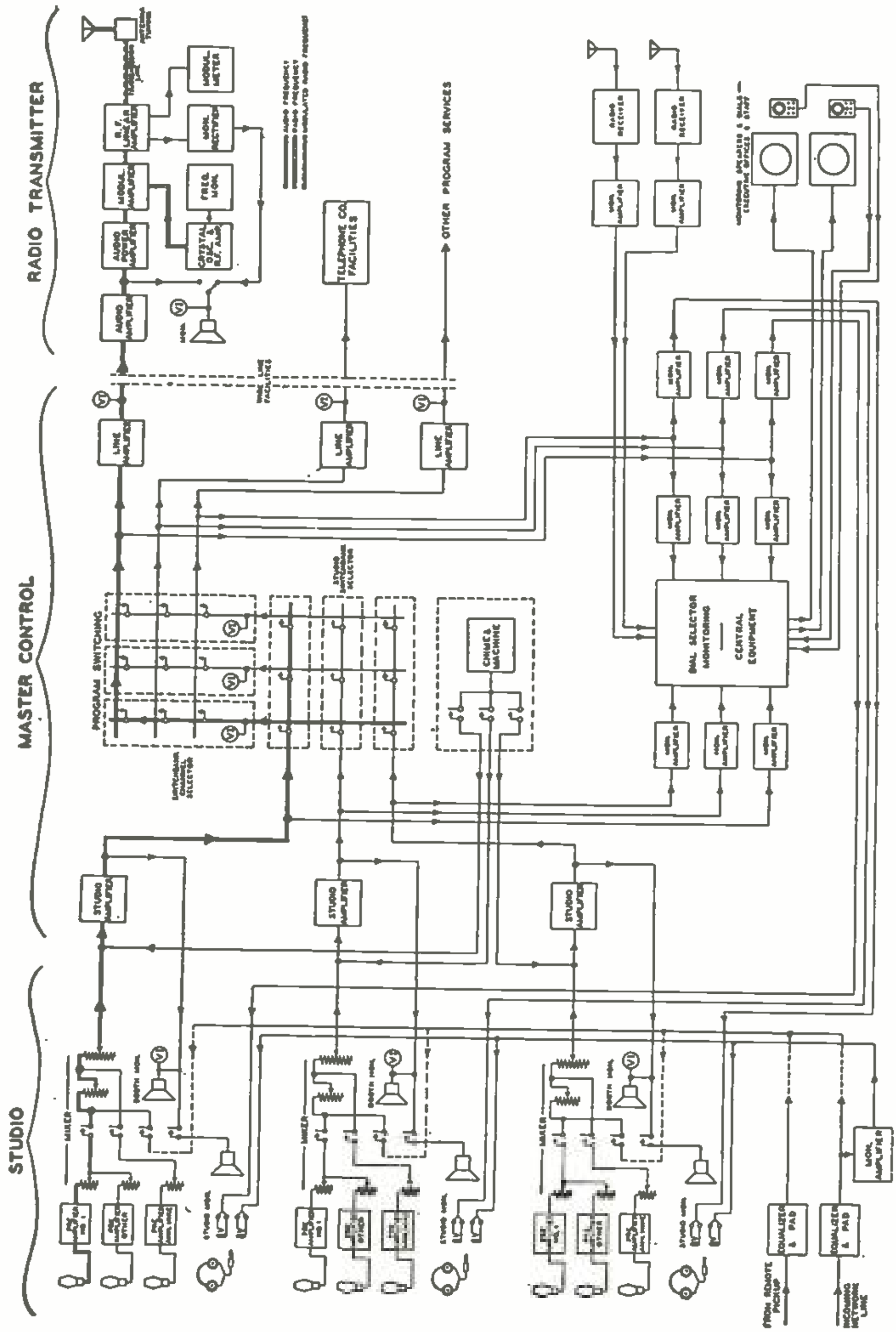
WHERE SPLIT SECONDS COUNT

In station setups where a comparatively large number of individual studios are involved, a central switching point known as "master control" is employed. Fig. 6-1 shows a simplified schematic of the NBC technical layout. This illustration shows how individual studios are connected through the switchbank selectors to the master control position, where the program or programs may be routed in any way desired. Fig. 6-2 is an illustration of NBC's Chicago master control.

The NBC program switching system is a standardized layout for all key stations of the network. Since more than one program is being handled at any one time, the setup must be as flexible and foolproof as technically possible. This calls for operation on a preset basis, eliminating as far as practicable confusion of switching a number of program sources in the split seconds allowed.

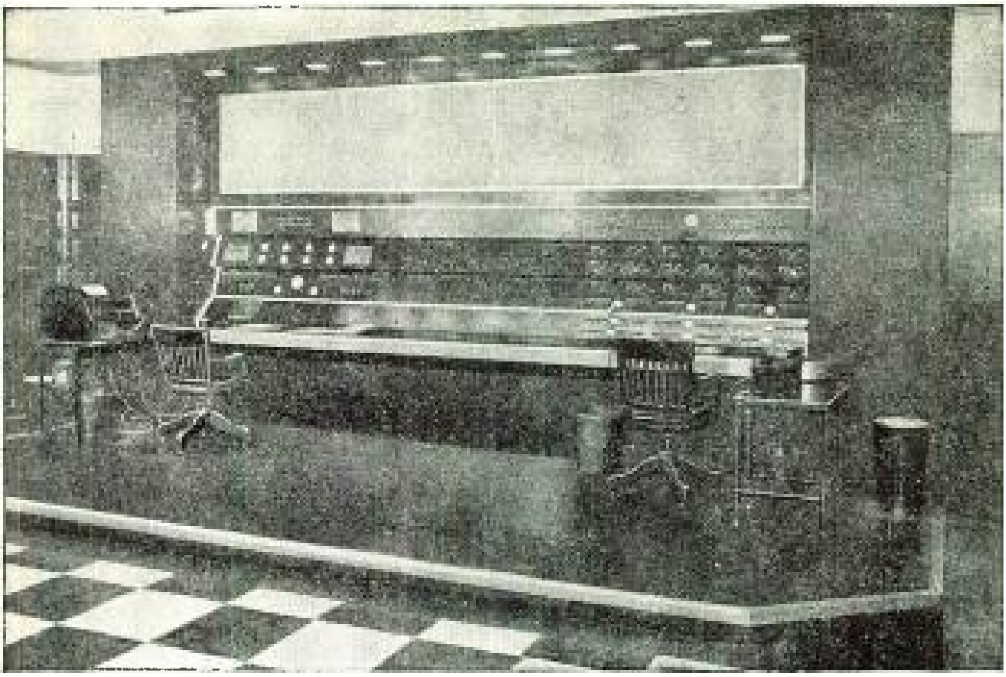
In the NBC system, the switchbank selector is a group of relays associated with outgoing channels, arranged for a single connecting means between any group desired and any single program input. A brief description of operation of the program switching system is as follows.

The program sheets or schedule sheets prepared in advance by the program and traffic departments indicate the program sources such as studio (and the particular studio number), for a remote point, or incoming network line. Also indicated are the outgoing channels feeding various points with the programs. At the start of operations, the channels required for each separate program are preset by operating numbered key switches in master control which are connected to separate switchbanks. Any switchbank is then connected to any program source at the proper time by operating the associated key switch on the switchbank selector panel, one panel being associated with each studio or other program source. This operation actuates the "carrier" lights at both the announcer's control desk in the studio and on the engineer's console in the control room, and is the signal for the pro-



Courtesy NBC

Fig. 6-1. Simplified schematic of the NBC layout, showing how individual studios are connected through the master control to the transmitter.



NBC Photo

Fig. 6-2. The master control at the NBC studio in Chicago.

gram to begin. In this way, programs are routed to the station's own transmitter and to the various other sources such as network, f.m., or special circuits such as international short wave.

Master Control of United Broadcasting Company

WHK master control is definitely not standard; it has grown through the years from the personal preferences of master control operators. From the foregoing statement it might seem to some that it would be a most glorious hodge-podge of gadgets by now. Actually it is the most simple and flexible system that could be imagined.

All program sources, each studio, the network, and four remote positions go to individual 12-tube repeater amplifiers. This provides twelve copies of each program source to be used for local station, network feed, f.m., monitor systems, etc. On the master control console are six banks of mechanically interlocking switches which route any program source to any or all of six line amplifiers. The interlocking is to prevent more than one key being depressed at any one time. The switchbank for WHK's line amplifier is different from the rest. Here, any program source can be put on any one of three faders, and interlocking prevents getting more than one program on any one fader. The faders in turn feed the line amplifier. The faders are considered necessary to smooth operation for getting in or out of programs late or early. There are no relays in any program circuit. One line amplifier feeds WHK's transmitter, one for the Mutual Network, one for an Ohio network, and three spares which can be used for anything. Also on the console itself are two switching panels which route any of 30 remote lines to any of the four remote positions. These same panels have facilities for line reversal and private telephones to each remote.

In addition to the console there are three banks of double depth relay racks which house power supplies, all program amplifiers, and monitor amplifiers. Every piece of audio equipment is brought out to normal through jacks so each one may be replaced or removed from the circuit in a few seconds in case of failure. All amplifiers have pads connected to their input jacks to take +8 vu down to whatever is necessary for that amplifier. This makes it possible to have every input and output +8 vu at 600 ohms. Even a shoemaker couldn't hurt anything by patching them all in series.

Function of Master Control Operations

Every individual station employing a master control has, of course, slightly different "rules and regulations" of procedure to suit their individual requirements and to satisfy the technical executives who are responsible for the co-ordination of all operations. The following rules of studio procedure that were compiled by the engineering supervisors of WBBM for guidance of their technical staff are presented here to acquaint the reader with general master control operations. The rules are divided into three sections: master control, studios, field. The letters *M*, *S*, and *F* have been used, respectively, to differentiate between these sections in numbering. Since all are closely tied in with the duties of the master control operator, they are presented here in their entirety.

MASTER CONTROL PROCEDURE

M1 Checking of New York Daily Operations Sheet

The master control engineer is required to check the routings of all network originations with the New York Daily Operations Sheet, and to compare the routings listed with those on the WBBM Daily Operations Sheet.

M2 Booth Check-in

For procedure to be used on a booth check-in, see *S5*. In addition, the Master Control engineer is to record on the WBBM Daily Operations Sheet, above the engineer's name, the time of the check-in. Should the studio engineer report someone as absent, Master Control should immediately notify the Program Department.

M3 Time of Making Preset

All relay presetups should be made approximately four (4) minutes before each regular switching period (see Glossary).

M4 Use of Switching Light

The signal for Master Control to switch from one studio (excepting studio M7) to another, *on all local originations,* will be the switching light.

M5 Checking Equipment

All equipment is to be carefully checked for gain settings, tube currents, etc., before being placed in service.

M6 Filling During Network Failures

For procedure see *S18*. In addition, on all scheduled stand-bys and all emergency fills to the network, Master Control is to patch the stand-by studio's cue speaker to the incoming network line.

M7 Cutting of Local Programs Running into Network Commercials or into Synchronization

All local programs running into synchronization or into network "musts" which are to be carried by WBBM, will be cut by Master Control only.

M8 Relieving of Engineers

a. A relief engineer is not to take over the Master Control if only (5) minutes or less remain before a switch. The engineer being relieved is to make the switch and see that the program or programs start properly.

b. An engineer is not to be relieved of duty until he has cleared the patching bays of all unnecessary cords.

M9 Checking Program Level

Master Control engineers are to keep a close check on the level of all programs, and to see that the proper level is maintained at all times. See also *S9*.

M10 Remote Check-in to Master Control

The Master Control should see that all remote engineers check in by twenty-eight (28) minutes before air time. If the remote has not been heard from by twenty (20) minutes before air time, provision should be made for a stand-by.

M11 Testing of Field Equipment

Remote engineers, before leaving the building for "pickups," are to give Master Control an audio test of their equipment. This equipment is not to be O.K.'d other than in good condition. The time of these tests is to be recorded on the WBBM Daily Operations Sheet, opposite the particular pickup, with the Master Control engineer's initials.

M12 Patching Up Remote Talk-Lines

Remote talk-lines are to be patched up only after the scheduled Studio engineer arrives in the booth and requests them.

M13 Recording of Inability of Studio Engineers to Get Program Procedure

All reports by Studio engineers of inability to ascertain the procedure on a particular program are to be recorded in the "penciled" comments, with the reasons and name or names of persons concerned.

M14 Making Setup for Following Morning

The engineer signing off each evening is to make the necessary setup in Master Control for the following morning, in order that the Studio engineer can put the station on the air should the Master Control engineer fail to arrive.

M15 Changes to the WBBM Daily Operations Sheet

All changes and notations to the WBBM Daily Operations Sheet are to be made in ink and initialed by the Master Control engineer.

M16 Signing of Engineering Department Copy of WBBM Daily Operations Sheet

Engineers must sign on the Engineering Department copy of the WBBM Daily Operations Sheet the time "ON" and the time "OFF" duty.

STUDIO PROCEDURE

S1 Ascertaining of Program Procedure

The engineer is required to acquaint himself with the procedure of all programs on which he is assigned. This is required regardless of the number of times the engineer may have been assigned to the show. If at any time it is impossible to secure this information, Master Control is to be notified in order that it may be entered in the "penciled" comments.

S2 Time of Check-in to Master Control

Studio engineers are to check in to Master Control not later than seven (7) minutes before air time.

S3 Remaining in Booth

Engineers are to remain in the booth between the time of checking in to Master Control and two (2) minutes after the program. It is, however, permissible to leave for the purpose of making a necessary last-minute change in the studio setup.

S4 Checking In for Rehearsals

Engineers scheduled on rehearsals are to report to the studio ten (10) minutes early, and have all equipment tests completed by rehearsal time. The failure of producer or talent to arrive does not relieve the engineer of this responsibility.

S5 The Check-in to Master Control

The check-in to Master Control is to be made as follows:

"*John Doe* checking in from studio three, T-H-R-E-E, for Columbia's School of the Air, *2:30—2:59½*, to SRR-NW-TC (give complete routing).

"The time is *2:20 and 40 seconds.* Woof!"

If it is a local program only, the engineer is to add after the routing that there *is*, or *is not*, a spot announcement following his program, and, if so, the studio in which it is scheduled.

It is the responsibility of both the Master Control and Studio engineer to make these check-ins carefully. The Master Control will repeat and spell out the studio number.

When making check-ins it will be understood that, *unless Master Control is informed otherwise*, all tests have been completed and *all* talent, including the announcer, is present.

S6 Time of Arrival in Booth on Remote Programs

Engineers are to arrive in the studio at least thirty (30) minutes before air time, if schedule permits, on all remote commercials and special events; on all other remote programs the minimum time is fifteen (15) minutes.

S7 Patching Up Remote Talk-Lines

Remote talk-lines are to be patched up only after the scheduled Studio engineer arrives in the booth and requests them.

S8 Studio Line-up with Remotes

The studio line-up with remotes should include the following, and preferably in the order shown:

1. Line and equipment test.
2. Level check (the Field engineer will call peaks).
3. Name and sequence of musical selections (if it is an orchestra).
4. Corroboration of air time.
5. Time check.

S9 A Proper Level to Be Maintained

It is the duty of the Studio engineer assigned to a program to maintain a proper level at all times. If the level from a remote is abnormal, correct it and then ask the Remote engineer to either raise or lower it.

S10 The One-Minute Warning

One minute before air time Studio engineers are required, first, to "kill" all microphones and give a one-minute warning over the "talk-back" mike to studio talent; and second, if a remote is scheduled, to give a one-minute warning over the telephone to the Remote engineer (see *F5*, a). After these warnings, the "talk-back" mike is to be disconnected.

S11 Remaining in Booth After Air Program

Engineers are to remain in the booth, and equipment must be left turned *on* for at least two (2) minutes after each air program.

S12 Use of Switching Light

The signal for Master Control to switch from one studio to another on all originations for WBBM *only* is the switching light. Hold the switch on until the channel light is removed.

S13 Turning On Cue Speaker

The WBBM cue speaker is to be turned on immediately after the one-minute warning preceding the start of the program for the purpose of ascertaining the time the channel will be received.

S14 Reporting of Trouble on Tests

All trouble encountered on tests preceding an air program must be reported to Master Control *immediately*.

S15 Filling Out of Program Report

An Engineering Program Report is to be filled out in full by the Studio engineer after *each* air program where there has been an interruption to the *normal* routine. This report is to be deposited in Master Control reasonably soon after the program.

S16 Disposition of Mikes and Cords After Each Program

After the completion of each program or a succession of programs from the same studio, all mike cords are to be rolled up and placed in a corner, and microphones are to be returned to their designated places in the Maintenance Department.

S17 Network Breaks: Length and Level of Sustaining Background

a. All regular closing network breaks and breaks used for split-network switching are of thirty (30) seconds duration, with the sustaining background faded out after fifteen (15) seconds.

b. All network breaks during a program which are used for station identification *only* are of twenty (20) seconds duration, with the sustaining background supplied for the entire period.

c. The level of the sustaining background during all CBS breaks is to be lowered to 30 (—10 vu).

S18 "Filling" During Network Failures

When standing by for the network or any portion thereof, should a failure occur or trouble develop which renders the program unintelligible, the "fill" should be made as follows:

First: In order to know *when* the trouble has cleared, turn on the CBS cue speaker, which Master Control has patched across the incoming line.

Second: Wait forty-five (45) seconds, and if by then the trouble has not cleared, fade off the incoming line and signal the stand-by announcer to make a courtesy; the "stand-by" should then be supplied.

Third: Immediately after the trouble has cleared on the cue speaker, the announcer should be signaled to make a second courtesy announcement rejoining the program.

Fourth: Fade out the "stand-by" and fade up the program.

In connection with the second and third items above, it should be understood that should a situation arise whereby an announcer is not available, the procedure remains the same with the exception that the courtesies are deleted.

S19 Procedure in Remote Program Line Failure

The following procedure is to be followed should a remote program line develop trouble:

If the trouble develops before air time, Master Control is to be notified, and both points (Master Control and Field) will then reverse lines. The remote engineer should then feed a test as usual. One minute before air time Master Control will disconnect the remote feed to the studio (to guard against a feedback), and supply "cue" over the substitute program line. The remote should then start the program *five* (5) *seconds after the proper cue,* which will be heard on the monitoring phones. This five seconds will be used by Master Control in normaling the feed to the studio. If the remote program consists of an orchestra which is to be announced from the studio, five (5) seconds will be allotted between musical selections.

The foregoing procedure is formulated, of course, on the assumption that the regular program line cannot be used even for cue purposes.

S20 Channel Lights: Taking Away of

Unless Master Control directs otherwise, the channel light or lights on *each* network origination will be taken away fifteen (15) seconds after *a*, the middle CBS cue (if any), which is used for split-network switching only, and *b*, the closing CBS cue. In the case of *a*, the channel light or lights will be returned in the regular manner (see *S21*).

S21 Signal Used to Begin Network and Local Originations

a. All regular network originations will start five (5) seconds after the channel light or lights are received.

b. All regular local originations will start *immediately* upon receipt of channel light or lights.

S22 Network and WBBM Remote Originations: Opening "Go-Ahead"

All originations for both the network and WBBM which open from local remotes will start by a verbal "go-ahead" from the Studio engineer.

S23 The Cutting of Local Runovers

On local "runovers," all cuts which are made necessary because of synchronization or network "musts" to be carried by WBBM, will be made by Master Control.

S24 Nonrelief Period of Engineers

When programs originate consecutively in the same studio, engineers are not to relieve each other during the last three minutes of a program and the first two (2) minutes of the following program, and then only after the relief engineer has familiarized himself with the routine of the program or programs.

S25 Turning Off of Equipment

With the exception of studio three (3) after 6:00 p.m., all equipment *is to be turned off* when it is not being operated by *a member of the Engineering Staff.*

S26 Studio Engineer Should Be Able to Put Station on the Air

Should the Master Control engineer fail to arrive for a "sign-on," the Studio engineer should be able to put the station on the air. For this reason, he should acquaint himself with the operation and setup of the following equipment:

1. Battery supply, and associated switches.
2. Local relay channel.
3. Patching of phonograph to studio.

See also *M14*.

S27 Use of Telautograph

Corrections to the WBBM Daily Operations Sheet will be written on the Telautograph. The engineer upon arriving in a booth should note all corrections affecting him, and change his own schedule accordingly.

S28 The Daily Work Sheet and Weekly Time Sheet

a. A Daily Work Sheet is to be filled out in full each day and deposited in the box provided in the Maintenance Department.

b. The Weekly Time Sheet, which is posted on the bulletin board in the Maintenance Department, is to be filled out each day showing the number of hours worked. This sheet is to be initialed by the engineer at the end of the week.

FIELD PROCEDURE

F1 Returning of Field Equipment

Field engineers are required to return equipment to the Maintenance Department after the engineer's last pickup for the day, and place it in its proper location.

F2 Doors That May Be Used When Taking Equipment In or Out of Building

Field equipment may be carried in or out of the Wrigley Building through any door *except the front door* up to 6:00 p.m. After this time, call for a building watchman to open a side door.

F3 Guests on Pickups

Engineers are not to take guests to remote pickups at any time.

F4 Testing Remote Equipment

Engineers before leaving for the field must give their equipment an audio and mechanical test as follows:

First: Check 1) microphones and cords for defects; 2) quality; 3) output level; 4) microphonics; 5) tube shields, observing that they are in properly, etc.; and 6) volume indicator.

Second: Recheck the first four foregoing items with Master Control.

F5 Check-in and Line-up from Field

a. Remote engineers are to check in to Master Control with an equipment test at least one-half hour before air time on *all* programs (see also *M10*). Leave a test on the line until the one-minute warning, which will be given over the telephone by the Studio engineer, and then fade out the equipment.

b. The line-up to the Studio engineer should include the following, and preferably in the order shown:

1. Line and equipment test.
2. Level check by calling peaks.
3. Name and sequence of musical selections (if it is an orchestra).
4. Corroboration of air time.
5. Time check.

On "3" above it is the duty of the Remote engineer to check carefully the name and sequence of musical selections with the orchestra leader.

F6 Lowering of Sustaining Music

On all remote orchestras which are announced from the studio, the Remote engineer is to lower the level of the sustaining between musical selections to 30 (—10 vu).

F7 Procedure in Remote Program Line Failure

See *S19*.

F8 The Daily Work Sheet and Weekly Time Sheet

See *S28*.

GLOSSARY

Channel Lights. The small circle of lights found in the center of each booth console and used as *the* signal which that studio is feeding an "outgoing" line. Each light represents a separate line.

Check-in. The verbal report by the Studio engineer to the Master Control that he or she (i.e., the Studio engineer) is in the booth for his or her next program (see *S5* and *M2*).

Cue Speaker. The small speaker located in the booth console which can be patched by Master Control to either the local or the network program.

Daily Log. See *Master Control Log.*

Daily Operations Sheet. See *New York Daily Operations Sheet.*

Daily Schedule Sheet. The schedule which shows the rehearsal time, air time, studio number, destination (i.e., network or local), and the name of the engineer assigned to each program. This schedule is posted on the bulletin board in the Maintenance Department each evening for the following day's operations.

Daily Work Sheet. A form which is to be filled out by each Studio and Remote engineer every day showing the name and time of all rehearsals and air programs worked.

Engineering Program Report. A report filled out by Studio engineers for the information of Master Control, and which explains any and all interruptions to the normal routine of an air program.

Master Control Log. The daily record of all abnormal operations kept by Master Control.

Network. The entire Columbia Broadcasting System or any portion thereof.

Network Break; CBS Break. The 30- or 20-second period at the end or in the middle of each program which is used for station identification. It always follows the words, "this is the Columbia Broadcasting System."

New York Daily Operations Sheet. The New York Daily Schedule which shows the exact network routing of each program.

One-Minute Warning. The warning that everyone should remain quiet, given to talent in the studio and/or a remote one minute before a program is scheduled to take the air.

"Penciled" Comments. The log kept by Master Control in which is recorded only material of a purely engineering nature.

Regular Program Schedule. See *WBBM Daily Operations Sheet.*

Remote. Any program originating at a point outside of the studio from which that program is controlled.

Split Network. A broadcast period during which time the Columbia Network is divided into two or more sections; i.e., more than one program is being originated at the same time.

Switching Light. A light located in Master Control and turned on by a switch on each booth console. It is used to signal the Master Control engineer to switch to the following studio or program.

Switching Period. Each quarter hour, i.e., the 15-, 30-, 45-, and 60-minute period.

Synchronization. The period after sundown each day during which time radio stations KFAB and WBBM broadcast simultaneously on the same frequency.

Talent. The person or persons associated with a program and who will be heard on the air during that program. This includes the announcer.

Talk-Back. The equipment used in speaking from the booth to the talent in the studio during rehearsals.

Trouble-Report Form. The report filled out by each engineer and filed with the Maintenance Department for each piece of equipment found defective.

WBBM Break; Local Break. The station identification.

WBBM Daily Operations Sheet. The official schedule of program operations for the day for WBBM.

Part 3

OPERATING OUTSIDE THE STUDIO

Chapter 7

REMOTE-CONTROL PROBLEMS

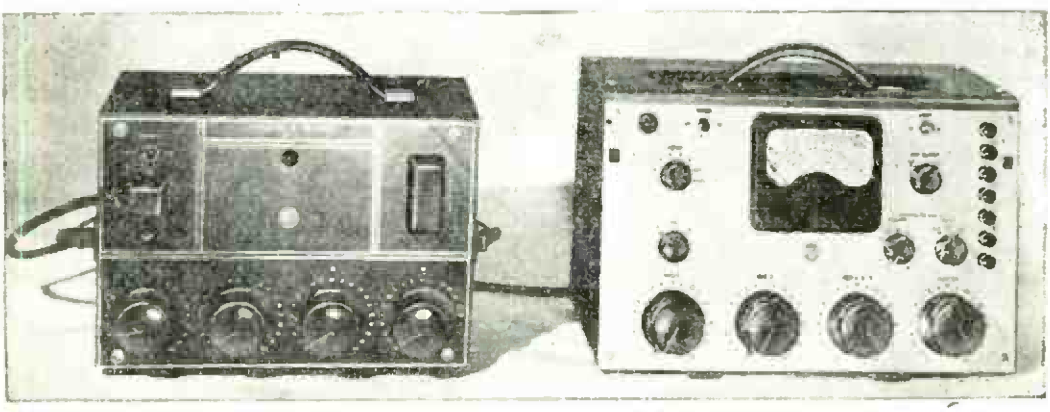
THE HISTORY of the development of radiobroadcasting since its earliest days, when the mere broadcasting of actual sound was miracle enough to create unbounded interest, has witnessed an almost fantastic evolution of technical equipment and technique of operation. Even during the earlier period when amplifier response and the old magnetic loudspeakers so limited the possible fidelity capabilities, broadcast engineers recognized the troublesome problems associated with the room or "studio" in which the program originated. Ordinary architectural construction did not satisfy the requirements for smooth control and faithful reproduction. This led to a detailed study and development of both architectural design and acoustical treatment to suit the needs of broadcasting. Although many experts believe that the final answer to this problem has not yet been found, they all concede that modern broadcast studios have spelled the difference between the success and the utter uselessness of high-fidelity amplifiers, microphones, and line or relay links.

Broadcast programs, however, are as varied as the interests of the more than twelve-million people who comprise the listening audience. It is inevitable that a great number of programs must originate at some point other than in a carefully designed studio with a permanent and complex studio control console and amplifier racks. Such programs as speeches and political rallies, news commentators "on the spot," audience participation shows from theatres or auditoriums, sports, religious programs from churches, popular music from night clubs, classical music from concert halls, and novelty and variety shows from theatre stages necessitate a special department at each broadcast station to handle such events adequately.

There are certain exacting requirements for remote-control equipment. The remote operator will encounter conditions that will be far from favorable for the type of program to be broadcast. If the specific location produces very decided effects, he must either use them to his advantage or avoid them. It is the purpose of Part 3 of this handbook to outline the general type of remote-control equipment, and to discuss comprehensively the problems encountered in the best utilization of this equipment to achieve the desired results.

Remote-Control Amplifiers

Equipment used for remote-control broadcasts must provide the same means of mixing the outputs of the microphones and sufficient amplifier gain for use of low-output high-quality microphones as does the main studio equipment. It is obvious, however, that the equipment must be conveniently portable and, therefore, limited in size and weight. For this reason, many of the earlier types of remote ampliplifiers used low-level mixing circuits requiring only one preamplifier tube for all microphone inputs. Recent advances in circuit design, however, have permitted the use of high-level mixing circuits with all its advantages, without adding to the bulk of the equipment. Power for this equipment is supplied by batteries or power-line current where available, or either, in case of power-line failure. Dynamotor supply is used in some mobile applications.



Courtesy RCA

Fig. 7-1. Typical equipment used for remote-control broadcast. A separate high-level mixer (left) is connected to the main amplifier.

Fig. 7-1 is a view of one type of remote equipment. The main amplifier to the right (RCA BN-2A) is a complete unit in itself, containing provisions for high-level mixing of three microphones, with a fourth which can be switched to channel 3. The illustration shows a separate 4-channel, high-level mixer connected to the main amplifier, allowing 7-channel operations.

The RCA BX-2A weighs only 29 lbs., complete with a-c cable and space tubes. A-c power supply is included, but batteries may be used by means of a battery cover pack. A line switch allows the program to be fed to the output channel and to a public-address amplifier, simultaneously, or the cue circuit may be switched to isolate the remote amplifier and feed the PA direct. The volume to the PA feed is on a separate control, allowing the remote operator a complete control of PA volume and feedback limitation.

Each input uses a balanced transformer with electrostatic shielding so that no special precautions are necessary in the grounding of microphones. (Discussed later in this chapter.) A 1620 tube is used in each of the preamplifiers. These tubes are connected with each mixer in parallel to feed the first stage of the higth-level amplifier. This stage employs another 1620 pentode connected with inverse feedback from the master gain control. The unique design of this arrangement produces maximum feedback with minimum gain, which reduces any inadvertent overloading of the first stage by announcers "blasting" the microphone or by excessive background noise. The second and third stages of the high-level amplifier each utilize a 6J7 pentode connected to the output transformer. Further inverse feedback is taken from the plate of the last stage to the second stage cathode. Each channel has an over-all gain of 92.5 db, more than adequate in any situation.

Since the level requirements for remote lines will vary with conditions, provisions are made for a multiplier arrangement on the vu meter so that zero reference may actually range from zero to +8 vu. On some open wire lines, for example, it may be advisable to feed a higher level to override line noise.

One example where this was found necessary occurred on a field broadcast handled from the Indianapolis Municipal Airport which consisted of interviews of incoming passengers from the planes. The remote amplifier was battery-operated to avoid any r-f feed-through from the control-tower transmitter via power lines. In spite of this precaution, the control-tower transmitter signal was feeding through so strongly on the broadcast line that the tower operator completely swamped the announcer when feeding an actual "zero-level" to the line. By adjusting the vu multiplier switch to +6, and peaking zero on the vu, the tower operator was down far enough below the program level that, when he came on with instructions to the pilots, he simply provided an "on the spot" atmosphere to the program without spoiling it entirely.

General Remote Operating Problems

The remote operator is faced with conditions so varied and complex that any discussion of a specific type of pickup must necessarily present only general principles involved.

A singer's voice is given a certain recognized timbre by the breath which carries the sound from vibrating vocal chords into the modifying air cavities of the head. As these sound waves emerge they disturb the air in the place of origin in all directions, but principally in the direction which the singer is facing. The microphone will pick up the sound anywhere in the room. Good transmission will depend upon the relationship of the position between performer and microphone, and also upon the relationship of position with the walls, floor, and ceiling of the room. The air cavities and acoustical condition of the air boundaries will affect the character or "timbre" of the sound just as do the air cavities of the singer's head which determines his original voice quality.

Thus it becomes apparent that the varied acoustical conditions encountered will place considerable importance on the type of microphone to be used and method of placement of the microphone. For example, the operator may find the surfaces bounding the point of pickup to be highly reflecting in character to sound waves, causing distinct "slaps" and echoes to be prevalent. This condition is caused by deflecting surfaces parallel to each other, and is the reason why "live-end" broadcast studios are constructed with no parallel surfaces existing. Under this kind of handicap, the operator must use the directional characteristics of the microphone to the best advantage. He could not, for example, use a bidirectional microphone with one live side toward the pickup and the other live side toward a highly reflecting wall.

Due to the nature of remote-control pickups, the microphones used are nearly always of the unidirectional type. This permits much better discrimination between wanted and unwanted sound, since the noise level at any remote point is quite high compared with a broadcast studio. The unidirectional characteristic is convenient to aid in preventing large amounts of reflected sound-wave energy from actuating the microphone elements. Since the intensity of a sound wave decreases as the square of the distance, increasing the distance between the sound source and the reflecting surfaces (where this is possible) will decrease the amount of reflected sound-wave energy at the microphone.

By experimenting with the distance between sound source and microphone, it may be observed that the relationship between original and reflected sound will vary over a considerable range. Thus by decreasing this distance a greater proportion of original sound is obtained, and by increasing the distance (between wanted sound source and microphone) a greater proportion of reflected sound is obtained. Music in particular needs a certain amount of reflected sound for brilliance and color. Too much reflected sound will cause a "hollow" tone and uncomfortable overlapping of succeeding musical passages. If the amount of reflected sound is too small, such as in many studios overtreated with sound absorbent material, the music will be lifeless.

Stray A-C Fields

Some of the older-type microphones and remote amplifiers using an unbalanced input (one side grounded) cause a considerable amount of extra work from certain locations where stray a-c fields exist near the microphone or run of mike cord. Fluorescent-light circuits in particular are hazards for the remote operator.

When a noticeable noise level is apparent upon a remote setup, and it increases with an increase in mike gain and stops upon disconnecting the mike, it is apparent to the operator that he must try reorientation of microphones and/or changing of the cord runs. Interference from fluorescent lights shows up as a raspy frying noise in the headphones.

In permanent installations where fluorescent lights are used, where microphones may be placed in close proximity, the transformers of the lighting fixtures are well shielded and often placed some distance from the mike area. Shielded wires then connect the fixture to the transformer.

At remote locations where the program is apt to be a "one-shot" affair, either the lighting fixtures must be moved or turned off during the show, or the microphone and performers must be moved until interference is eliminated.

Latest microphones and remote-amplifier equipment gives negligible trouble in this respect due to improved shielding in microphones and input transformers, as well as "balanced-to-ground" circuits.

Chapter 8

REMOTE VERSUS STUDIO PICKUPS

THE PROBLEM of broadcasting concerns the transmission and reception of voice and music with the preservation of all the original values. This precludes that any effect should be added to or withdrawn from the original intent. In radio, sound can play on the emotions of the listener only as effectively as the transmitter and receiver equipment, studio conditions, and the skill of the engineers will permit. Microphones and amplifiers are today of such good quality that no practical limitations to true fidelity exist from mechanical or electrical characteristics. Modern broadcast studios are such that only slight limitations exist for faithful transmission of sound. This emphasizes, insofar as remote-control broadcasting is concerned, that the skill of the engineer or producer responsible for microphone setups and operating technique, is of utmost importance. This becomes doubly important when it is realized that each orchestra of any type has its own identifying qualities resulting from instrumentation, musical technique, and conductor's interpretations, all under the influencing factor of microphone placement and acoustical conditions of the point of origin.

The effects desired by the orchestra conductor may be achieved only by proper relationship of the microphones to the musical instruments. This "proper relationship" is directly influenced by the acoustical condition of the pickup area. For transmission of pure musical tones of a violin, the microphone must be far enough away from the sound holes of the violin that the reflected sounds may be caught in all their beauty denoting rich and true harmonic content. Conversely, when special effects are desired such as in many instrumentations of rumbas in dance orchestras, the microphone should be so near the violin as to bring out the harshness of the resined bow drawn across the strings of the instrument.

General Comparisons of Studio and Remote Pickups

Perhaps the most striking difference between studio and field pickups is the complete lack of permanent facilities of any kind in the field. The Bell Telephone System installs a "broadcast loop" upon order from the program or traffic department of the station. Sometimes two loops are installed, one to be used as a "talk" line direct to the control room at the studio, or for emergency broadcast service in case of trouble with the regular broadcast line. These lines, however, must be installed as conveniently as possible to the source of the broadcast, yet as inconspicuously as can be arranged. For this reason, it is often a matter of a "line hunt" on behalf of the field engineer, and this is one reason why he arrives at the remote point a long time in advance of broadcast time. The line or lines may be found under tables, behind chairs, piano, organ console, or what-have-you on the stage, or it may be in a room off from the main room where the broadcast is to take place. It is usually tagged with an identifying card such as "WIRE Broadcast."

Since the problem of good transmission of talks or speeches at remote points is not nearly so difficult as that of musical pickups, the discussion to follow will concern music. Musical programs may originate at such places as ballrooms, restaurants, night clubs, and cafes featuring dinner music, and music for dancing and floor shows. The situation calls for a decided difference in technique of technical production between studio and remote broadcasts.

In the ideal studio musical setup, only one microphone is used at sufficient distance, with the musical instruments grouped and positioned so as to blend into the proper balance at the microphone position. This procedure not only simplifies the problem of control, which always makes for a better effect, but also leaves the problem of orchestral balance in the conductor's hands where it rightfully belongs. Multiple microphone arrangement will place the maximum responsibility for balance of the various sections in the hands of the operator mixing the outputs of the various microphones.

At remote points, however, where so much activity such as dining or dancing occurs, microphones must be placed close to the musicians. This is inevitable since otherwise the background noise would result in a disagreeable hodge-podge of confusion. This close microphone arrangement calls for the use of more than one microphone to achieve the desired balance; otherwise the instruments closest to the microphone would dwarf the rest of the orchestra. Then the setup is divided into units of like instruments or combination of instruments, each unit being covered by a separate microphone so that the volume from each unit may be adjusted at the mixing panel to achieve the desired balance.

The practice has some advantages for remote-control pickups other than avoiding background noise. Acoustical conditions that might severely affect the broadcast are minimized to the fullest extent, since the ratio of any reflected sound to the original sound is small. Then too, although some loss of tonal brilliance results from close microphone arrangement, good instrumental definition is gained, which is important for dance broadcasts.

Symphony music and church broadcasts are different in this respect in that the audience is comparatively quiet, and the pickup may be treated more as a studio show by studying the acoustical conditions existing at the point of origin.

Chapter 9

REMOTE MUSICAL PICKUPS

An observation of Fig. 9-1 will reveal the principles involved for a typical dance orchestra broadcast. Insofar as the operator is concerned, this setup divides the orchestra into three separate units: microphone #1 for saxophone and clarinets; #2 for trumpets, trombones, and soloist; and #3 for string bass and piano. Microphone #3 is very handy for special emphasis of the rhythm section, or piano or string bass solo passages. It will be noted that when the trumpets are open, they are behind the trombones and caught on microphone #2; when muted, they step down ahead of the trombones and immediately in front of the microphone. Muted trumpets or trombones must be played with the muted bells very close to the face of the microphone. The same is true of any wind instrument upon which the player is producing subtones. The subtones of any wind instrument are just as low in volume, even though open-belled, as the softest muted instrument. This, then, calls for close co-operation between the conductor and his musicians and the engineer responsible for proper pickup. Many times, important solo "licks" of a particular passage may be lost by lack of co-ordination.

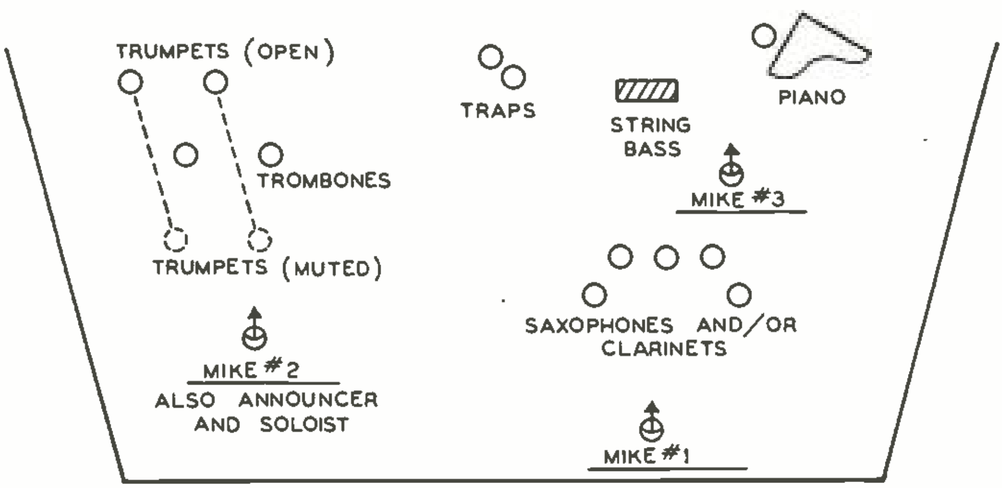


Fig. 9-1. Seating arrangement of dance orchestra and placement of microphones for a typical remote pickup.

Brass Bands

Although the 4/4 type bands share comparatively small time in radio, their particular peculiarities pose special problems in pickup. A number of community organizations, fraternal societies, and, of course, the armed services participate in radio through presentation of brass bands. These pickups very often must be made out of doors, the least favorable spot for broadcasting. With no outdoor shell or walls of any kind, no reflection of sound can occur to create the ideal polyphased sound dispersion so important to broadcasting technique. Under these conditions it is again necessary to use multiple microphone pickups, grouping the units by means of spotting separate microphones where needed as determined by trial.

For a fair-sized band organization, the units are usually as follows: one microphone for the clarinets, piccolos, and flutes; one for the English horns, bassoons, bass clarinets, saxophones, and tubas; and one for the French horns, trombones, and trumpets. The tympani, traps, and chimes are usually placed in the lower sensitivity zone of one of the microphones which prevents the use of excessive distance for proper balance. Indeed, the sensitivity pattern characteristics of the particular microphones used must be thoroughly understood for any kind of musical pickup. Tympani, when used with brass, are very predominate in character when placed in an equal sensitivity zone to the rest of the instruments. Just the opposite is true when they are used with strings, since the masking effect due to the characteristics of the musical instruments themselves tends to subordinate the tympani sound.

When well-designed outdoor shells are used, the ideal condition exists for brass-band broadcast. Usually only one microphone is used, suspended some 15 feet out and above the front-line musicians. As before, predominate instruments, such as tympani, traps, and chimes, are placed at the side in a lower sensitivity area of the microphone.

Salon Orchestra Remotes

Some dining places have salon or chamber music organizations which are picked up for broadcasting during the noon or early evening hours. Since a salon orchestra's library concerns the more serious type of music with many low passages, precautions must be taken to subdue as much as possible the noise of the patrons. An intimate microphone placement is therefore indicated.

Usually the salon group is small, ranging from string trios and quartets to about ten members. For the smaller groups, one microphone raised quite high and slanted down at an angle of about 35 to 45 degrees with the floor will be adequate. A hard floor with no covering will aid in obtaining just the amount of brilliance necessary for this type of pickup. A salon orchestra requires more definition than brilliance in musical tones.

Symphonic Pickups

Symphony orchestra programs have become a regular feature on the air each season and quite often must be broadcast from a remote point rather than from a regular broadcast studio. Thus far, musical setups have been discussed involving a comparatively small number of musicians and a specific type of instrumental structure. The symphony orchestra, however, is many orchestras in one. The engineer is concerned with the proper grouping of four distinct instrumental sections:

1. *Strings:* violins, violas, cellos, string basses.
2. *Woodwinds:* clarinets, bassoons, English horns, flutes.
3. *Brasses:* trumpets, trombones, French horns, tubas, euphonium.
4. *Percussions:* snare drums, bass drums, tambourines, triangles, cymbals, piano, harp, xylophones, marimbas, tympani.

To this instrumental setup, vocal soloists and choirs are often added, as for Beethoven's "Ninth Symphony" or Verdi's "Requiem." The musical score itself will influence many times the necessary spotting of microphones. For such numbers as the delicate "Clair de Lune" of Debussy, the perspective of the violin passages should be distant, with a rich and brilliant tonal quality. In numbers such as the Strauss waltzes, the perspective of the strings should be closer and more strident in character. This problem will be outlined in more detail presently.

As a general rule, the arrangement of the symphony orchestra for broadcast is the same as for a regular audience performance. The instruments vary in volume of sound produced and therefore in penetrative quality. Strings produce the least volume, then flutes, clarinets, horns, trumpets, and percussion instruments.

The acoustical situation for symphony broadcasts is generally better than for most other remote controls since the auditorium is usually designed for such large groups and made compatible with good listening for the audience, although not always ideal for broadcasting.

It is easier from a good transmission standpoint to encounter an auditorium that is too "live" and reverberant so that wall, ceiling, and floor treatments may be added, than to start from one that is too "dead" to sound reflection.

The correct setup for a symphony orchestra is always arrived at on the first rehearsal by trial and error. A number of microphones are spotted at the most likely points so that each may be tried without the commotion of continually moving one microphone. The most likely setup is one microphone suspended at a height of about 15 feet about 20 feet in front of the violins. A separate microphone must be used for vocal solos, since a closer relationship of vocalist to microphone must prevail for proper balance.

A typical setup for a full symphony orchestra was shown in Fig. 4-7 and no difference need occur for remotes. Some deviations occur in practice with various symphony orchestras. Toscanini's NBC pickup, which originates in a regular studio, uses two microphones for the main orchestra. Due to the directional characteristics and angle of placement (one for each side section), the orchestra is effectively divided into two microphone fields with little overlapping. Sometimes another microphone is suspended directly over the violin section for special effects on certain compositions as mentioned before. The Ford Sunday Evening Hour, broadcasted over CBS on Sunday evenings, used two microphones on the choir for clarity and definition of diction.

In chapter 4 is a complete description of a specific symphony setup.

Church Remotes

Programs from churches usually involve both music and the sermon.

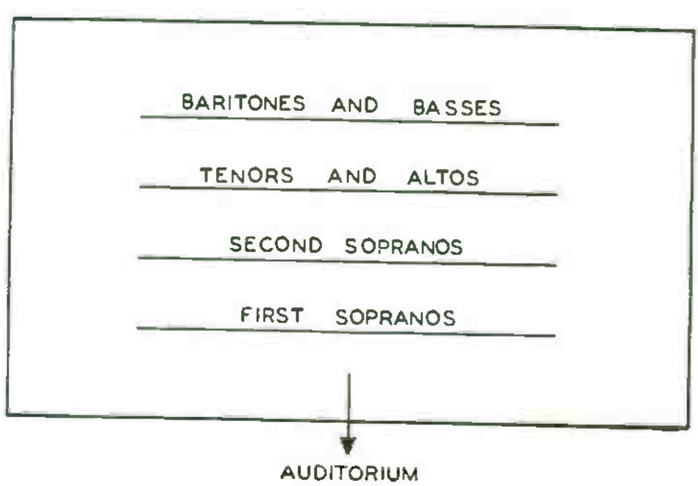


Fig. 9-2. The usual positions of a choir as shown here, results in too strong soprano response and insufficient alto and bass. Compare Fig. 9-3.

This ordinarily requires only one microphone when the minister's podium is directly in front of the choir as is the most common church arrangement. When vocal solos occur during the choral rendition, a separate microphone is necessary for proper pickup and balance. It will be noted in nearly all instances that, during solos being picked up by a microphone very close to the choir loft, organ accompaniment must be brought up to the proper background level by use of the rostrum microphone or microphone farther out in the congregation. This is due to the acoustical properties which are evident in nearly all churches causing the organ tones to be much more predominant out in the congregation than up near the choir.

Conventional choir arrangements are often not practical for broadcasting whether in a regular studio or at a remote point. Fig. 9-2 illustrates the usual arrangement of a choir as used for auditorium or church presentation. On a broadcast, this arrangement nearly always results in a predominance of soprano voices with very little alto or bass. Fig. 9-3 shows an arrangement much more satisfactory for broadcast purposes, resulting in a better all-around balance of voices.

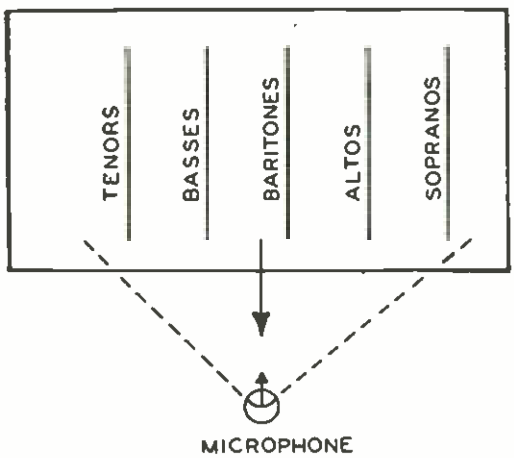


Fig. 9-3. By placing a microphone in this relationship to the rows of singers, a better all-around balance of voices is obtained over the arrangement shown in Fig. 9-2.

Although it would be impossible to cover all the details and complexities of remote-control pickups in a single discussion, it is hoped that the picture here presented has set forth the fundamental procedures that would help in a general way to approach a remote-control problem properly. To present an absolutely complete picture would be impossible, since acoustical conditions and orchestral intent varies as the number of places from which a broadcast can originate, and the number of different musical combinations existing. A good understanding of equipment and acoustical variations, however, will enable any engineer to achieve good results on this type broadcast.

Chapter 10

EYE-WITNESS PICKUPS AND MOBILE TRANSMITTERS

THERE ARE many types of events of wide public appeal that cannot be adequately covered by the usual methods of remote-control pickups using wire lines for links of communication. Among these are various kinds of sports such as boat racing, cross-country events, and golf matches. Aside from these events, there are the inevitable times of disaster such as floods, fires, earthquakes, and the myriad types of catastrophes that wreck ordinary communication services for many miles around the point of trouble. In order to be prepared to bring eye-witness accounts of happenings of these kinds to

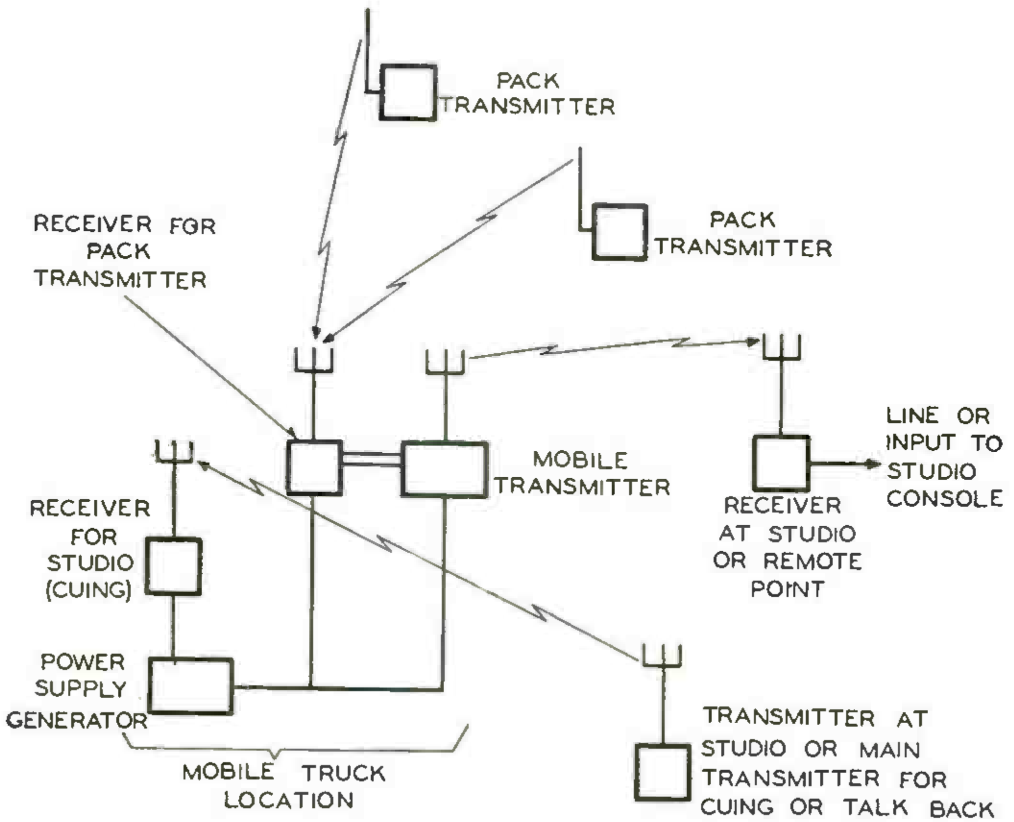
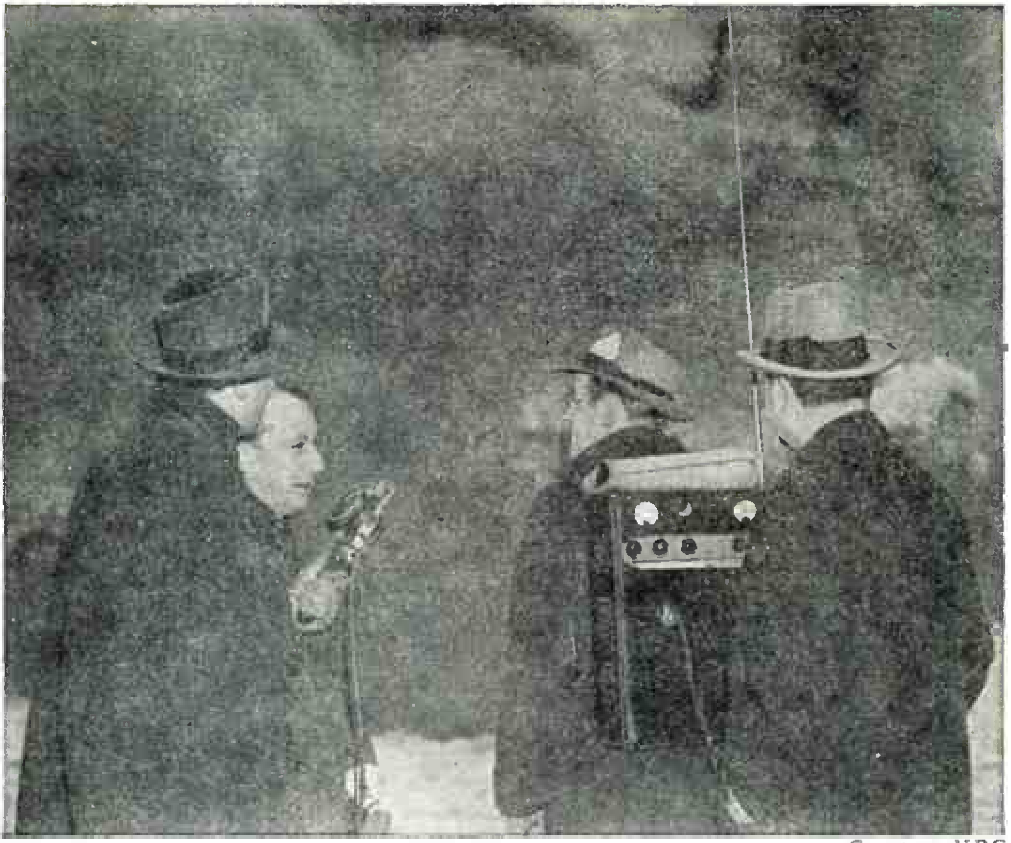


Fig. 10-1. Block diagram of equipment for pack-to-truck and truck-to-main transmitter relay transmissions.



Courtesy NBC

Fig. 10-2. Broadcasting an eye-witness account of the burning of the S.S. Normandie from an adjacent pier by means of a pack transmitter.

the thousands of interested listeners, most stations are equipped with portable and mobile relay facilities that utilize power supplies independent of utility companies, and also independent of any necessary wire lines, for relaying the signal to the studio or main transmitter.

There is probably no other division of radiobroadcasting that differs so radically from one station to another as the mobile-relay department. Fundamentally, however, the necessary inventory of equipment includes small portable transmitters known as "pack transmitters," a mobile transmitter and antenna mounted in a truck, receivers for cuing and pickup of pack transmitters, and power supplies for the equipment used.

Fig. 10-1 shows the fundamental layout of equipment used to broadcast any event as mentioned in the beginning of this chapter. Pack transmitters are low-output transmitters (usually about 2 watts) such as illustrated in Fig. 10-2. These transmitters are usually good for line-of-sight transmission only and therefore are picked up on a receiver in the mobile truck and fed to the main mobile transmitter. Mobile transmitters with their associated antenna systems are mounted in trucks or cars, such as that illustrated in Fig. 10-3.

Courtesy WOR

Fig. 10-3. The second link in an on-the-spot broadcast is often a mobile short-wave transmitter installed in a truck.

RCA BTP-1A BROADCAST TRANSMITTER

Description

The RCA type BTP-1A radio mike broadcast transmitting equipment, as shown in Fig. 10-4, is essentially a portable microphone designed for use in those applications where complete freedom from interconnecting cables is required. The maximum satisfactory range of the BTP-1A is determined largely by the interference level at the receiver location. Under ideal conditions several miles may be covered; however, building structures or other media possessing a high signal attenuation will limit the working range. Over-all transmitted quality is comparable to a network line.

The complete unit is carried by the operator and houses crystal microphones, crystal-controlled transmitter, modulator, and battery. The antenna is mounted on top of the unit. Handles are provided on both sides of the housing to permit carrying the unit in either or both hands.

As noted on the schematic diagram, Fig. 10-5, battery-powered tubes are used throughout. One section of a twin-triode tube is used

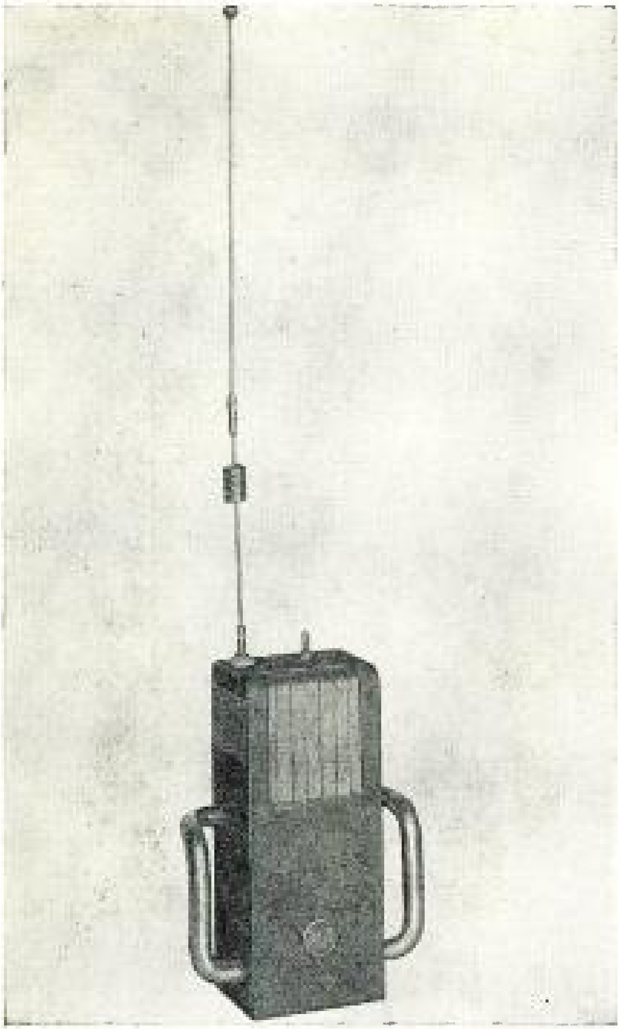


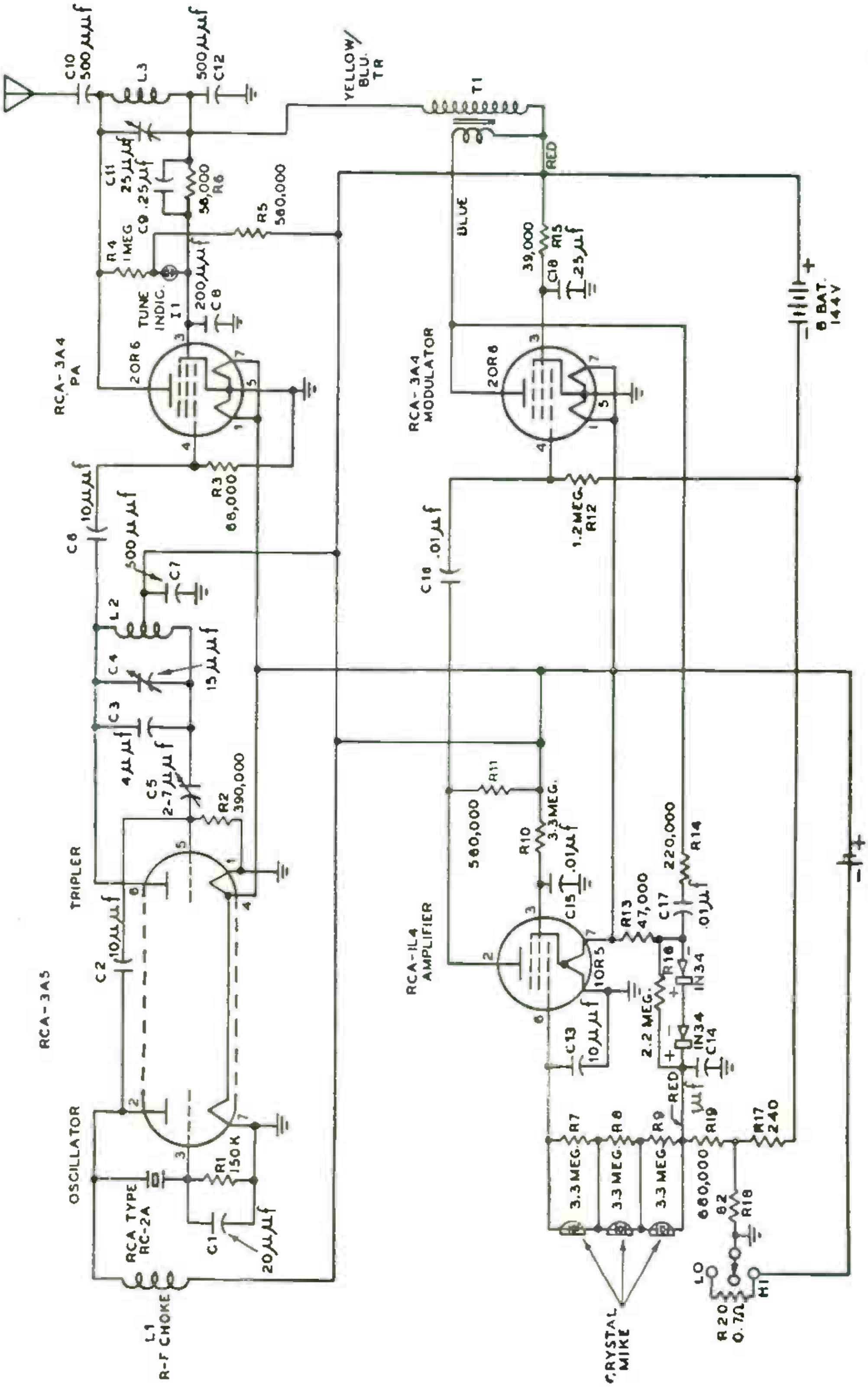
Fig. 10-4. A radio mike broadcast transmitting unit. It contains a microphone, transmitter, modulator, and battery. The antenna is mounted at the top. Under ideal conditions, the unit has a range of several miles.

Courtesy RCA

in a Pierce oscillator circuit with the second section of the tube being used in a tripler circuit. A pentode-type tube is used as a power-amplifier.

Three miniature crystal microphones connected in series, are used for signal pickup and are wired directly into the grid circuit of the first audio stage. *R-C* coupling is used between the first audio stage and the modulator tube. The audio output voltage of the modulator is transformer-coupled to the screen grid and plate of the power-amplifier tube. Overmodulation of the screen grid in conjunction with normal modulation of the plate, results in higher modulation capability, with less battery drain.

Overmodulation of the output is automatically prevented by rectifying the audio signal voltage appearing at the plate of the modulator and using the resultant d-c voltage to control the bias of the first audio tube.



Courtesy RCA

Fig. 10-5. Schematic diagram of the RCA BTP-1A broadcast transmitter pictured in figure 10-4.

The only control on the unit (except the tuning and neutralizing capacitors which, once adjusted, are not disturbed) is a three-position toggle switch mounted on top of the case. When in the vertical position, the battery filament circuit is open. When operated toward the left of the unit (as viewed from the front) the equipment is operating on reduced power. When the switch is operated toward the right of the unit, the equipment is operating on full power.

A neon lamp connected in the screen-grid circuit of the power amplifier tube serves as a power ON/OFF and tuning indicator.

Tuning

Caution. Before power is applied to this unit, licenses required by the Federal Communications Commission must be obtained.

1. Operate the toggle switch to apply HIGH power to the transmitter. The neon lamp will glow when the power is on.

2. With an insulated screwdriver, adjust the lower capacitor (*C4*, accessible through the lower opening in the upper-rear corner of the case) for minimum illumination of the neon lamp.

3. With the screwdriver, adjust the top capacitor (*C11*, accessible through the upper opening in the upper-rear corner of the case) for maximum illumination of the neon lamp.

4. Repeat steps 2 and 3 to obtain maximum illumination of the neon lamp.

An alternative method is to use an accurately calibrated receiver tuned to the desired operating frequency, using the *S* meter of the receiver as an indicator while making the preceding adjustments.

The neutralizing control, *C5*, is set at the factory and usually will not require readjustment unless changes have been made in the tripler stage components. When such adjustment is required, proceed as follows:

1. Make certain that the circuits are tuned to resonance.
2. De-energize the unit, then open the hinged sides.
3. Remove the crystal holder.
4. Connect an external milliameter in series with the +144-volt lead or inductively couple an oscilloscope to the antenna.
5. Energize the unit, then, using an insulated screwdriver, rotate the neutralizing capacitor *C5* to the position at which oscillation begins. Oscillation will be evidenced by a decrease in total battery drain or by r-f on the oscilloscope.

6. Rotate the neutralizing capacitor to the position at which oscillation stops.

7. Rotate the tripler tuning capacitor *C4* throughout its range and if oscillation is present at any position repeat step 6.

8. De-energize the unit, then remove the external meter, replace the crystal holder, and close the hinged sides.

Operation

1. Operate the toggle switch to either HIGH or LOW power as desired.

2. Speak into the microphone, using the unit in the same manner as any other type of microphone.

3. To remove the unit from operation, set the toggle switch handle in the vertical position.

Caution. Care must be exercised to protect the microphone as much as possible when using the unit outdoors, especially where inclement weather is encountered.

Maintenance

Very little maintenance is required to keep the unit in good operating condition. Use a blower at least once a week (when used indoors) and more often when used outdoors to keep the unit clean and free of dust.

TABLE 10-I

TYPICAL VOLTAGE MEASUREMENTS

Tube	Function	Pin	Plate	Pin	Screen	Pin	Grid	Pin	Filament
3A5	Oscillator/Tripler	2	128	—	—	—	—	4	1.5*
3A4	Power Amplifier	2-6	126	3	37	4	—9.6	1-7	1.5*
1L4	A-F Amplifier	2	51	3	15	6	—1.5‡	7	1.5*
3A4	Modulator	2-6	125	3	70	4	—9.0†	1-7	1.5*

* ±5% of rating.
‡ Measure from bottom of *R19* to ground.
† Measure from bottom of *R12* to ground.

A log should be kept of the operating time of the unit so that failure due to a weak battery may be prevented. A fresh battery will furnish power to the unit for four hours when operated continuously, and for eight hours when operated intermittently. It is highly desirable that spare tubes and batteries be carried by the operator, thus insuring minimum delay in the event of a failure during the broadcast.

Improper operation, as evidenced by distortion, or no signal output may be caused by a weak battery, defective crystal or tube, or incorrect tuning adjustments. Table 10-I is furnished as an aid in troubleshooting. All voltage measurements are made to ground.

Remote Broadcast Frequency Assignments

Remote broadcast pickup transmitters are assigned to the following frequencies, with certain limitations as noted:

26.10 to 26.48 Mc (nineteen 20-kc channels)

152.84 to 153.38 Mc (nine 60-kc channels. These channels may be assigned to remote pickup base and mobile stations on the condition that harmful interference will not be caused to the industrial radio services which share these channels.)

166.250 to 170.150 Mc (shared by government services in the 162 to 174 Mc bands. The sum of the bandwidth of emission and tolerance not to exceed 60 kc. Broadcast services are for Continental United States only, except within the area bounded on the west by the Mississippi River, on the north by the parallel of latitude 37°30′ N., and on the east and south by that arc of the circle with center at Springfield, Ill., and radius equal to the airline distance between Springfield, Ill., and Montgomery, Ala., subtended between the foregoing west and north boundaries. Frequencies will be assigned to broadcasters only on condition that harmful interference will not be caused to government stations, present or future, in the government band, 162 to 174 Mc. The use of these frequencies by remote pickup broadcast stations *will not* be authorized for locations within 150 miles of New York, N.Y.)

Technical Rules for Remote Broadcast Services

As ruled by the FCC, bandwidth for a-m phone on the above channels is 8 kc; for f-m phone, 40 kc. The specified band shall contain those frequencies upon which a total of 99% of the radiated power appears, extended to include any discrete frequency upon which the power is at least 25% of the total radiated power. Any emission appearing on any frequency removed from the carrier frequency by at least 50%, but not more than 100% of the maximum authorized bandwidth shall be attenuated not less than 25 db below the unmodulated carrier. Spurious or harmonic emission on any frequency removed from the carrier by at least 100% of the maximum authorized bandwidth shall be attenuated below the unmodulated carrier by not less than:

40 db with maximum plate power input to the final r-f stage of 3 watts or less

60 db with more that 3 watts and including 150 watts

70 db with more than 150 watts and including 600 watts

80 db with more than 600 watts.

Deviation due to modulation on f-m transmitters must not exceed plus or minus 15 kc from the unmodulated carrier. Maximum plate power input to the final r-f stage (a.m. or f.m.) shall not exceed 500 watts on the 26.10 to 26.48 Mc range, and 600 watts on the higher-frequency assignments.

The operator must ascertain that the transmitter is within 0.01% of the assigned frequency in the bands 26.10 to 26.48 Mc., and within 0.005% in the 152.84 to 153.38 Mc, and 166.250 to 170.150 Mc bands. Frequency stability on the 450 to 452 Mc bands are specified when the FCC authorizes such license.

The operator must also be certain that the call letters of the remote station are announced at the beginning and end of each period of operation (whether rebroadcast on the main station transmitter or not) and at least once every hour during the operating period.

Chapter 11

FIELD FACILITIES

When a program is to originate at some remote point from the regular studios, a number of facilities must be considered aside from actual equipment taken to the remote broadcast. Field facilities include the remote amplifiers, headphones, isolation coils, portable mike stands, equalization circuits, cue-in and cue-back facilities, etc.

Provisions that must be provided for proper check-in and co-ordination between studio and remote points will be taken up first.

Remote Keys at Studio Control

A remote broadcast point must check in to the control position before each broadcast, and also be able to receive the cue-back from the main studios in order to receive the proper cue for start of the remote broadcast. Although some stations, particularly when feeding a network of stations, use two separate lines for this purpose with a "PL" phone (private line) connected to the talk circuit, most stations now use a single line for broadcast and cue-back.

There are three ordinary positions of a remote key at the studio console which enable proper checking and co-ordinating functions to be performed. These are:

1. *Override.* This means that the key is connected to the proper remote line pair number and is in a position so that when the remote operator calls in he will be heard over the monitor system. In some cases a separate amplifier and speaker is used so that the remote operator check-in does not come in over the regular monitoring system which must, of course, be simultaneously monitoring the program signal.

2. *Cue and Talk.* In this position a microphone in the control room, or a telephone handset, may be connected to the line so that the operators may talk to one another for test of the line, time checks, etc.

It may also be terminated from a bridging connection of the monitor lines, so that the program at the studio may be fed back over the line for purposes of cueing.

3. *Broadcast.* In this position, the line is terminated into the mixer control which feeds to the program bus. The remote point is thus fed through the control console in the orthodox manner.

Typical Remote Key Circuits

Fig. 11-1 is a simplified schematic of one general type of remote check-in, cue-in, cue-back arrangement. Key *A* is a three-position remote key, and is the advance selector circuit for keys *B*,*C*, and *D* for whatever function that is to be exercised.

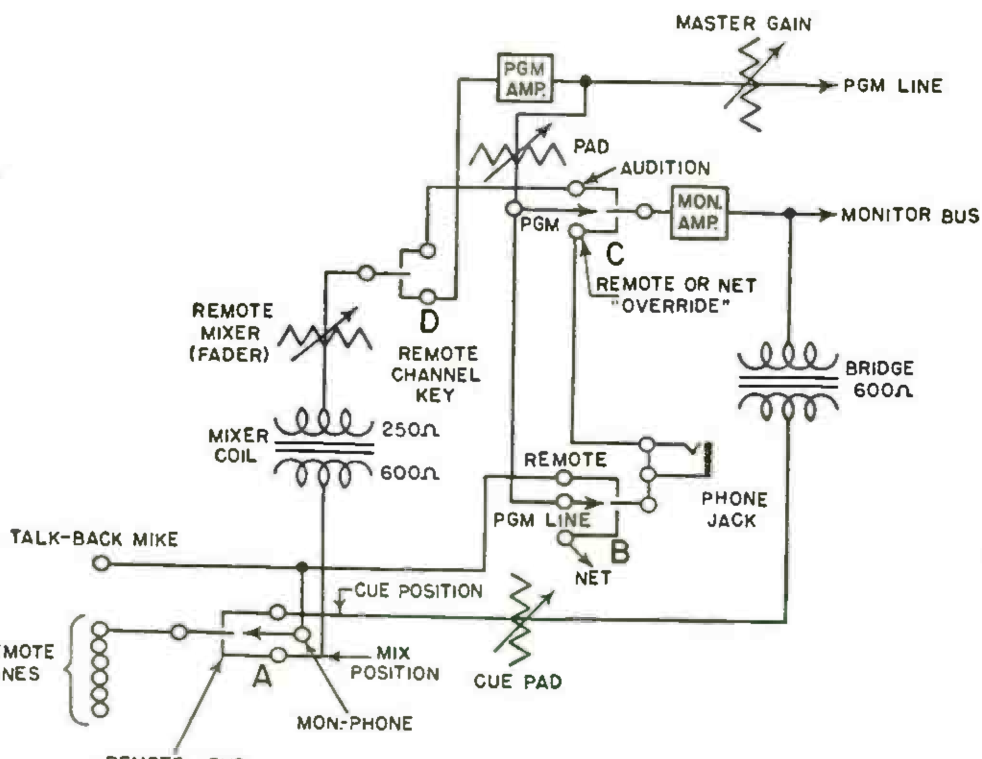


Fig. 11-1. Simplified schematic of a typical check-in, cue-in, cue-back remote key circuit. This circuit permits complete control coordination.

For example, when *A* is in neutral (the monitor-phone position) and key *C* is in "override" position, the remote operator will be heard over the monitor amplifier and speaker. This could also be fed into an auxiliary amplifier and speaker if desired. Key *B* is the headphone selector key, which may be placed in the "remote" position when desired to talk back to the remote point. To talk back, the remote key *A* is left in neutral, the phone jack key *B* is placed in "remote" position, and the phone or mike circuit actuated to talk to the remote point. Usually a "talk-back" button is used to connect the microphone, and disconnect the speaker by relay action. When desired to feed the "cue-back" signal over the line, remote key *A* is placed in the upper, or "cue" position, where the signal from the monitor amplifier is bridged over through the adjustable pad into the remote line. When going on the air, key *A* is simply thrown "down," or to the mixer input position, and the remote channel key *D* thrown to the "program" position. It is also noted that the remote signal may be fed into the monitor amplifier when key *A* is in "down" position by throwing key *D* to "audition" position.

Thus we have a complete control-co-ordinating facility for remote controls. Although the variety of arrangement and components used in practice is great, the general principles remain the same as outlined above.

Simplex Control of Remote Amplifiers

Many times it is highly desirable from an efficiency point of view to allow turning the remote amplifier on and off from the studio. This

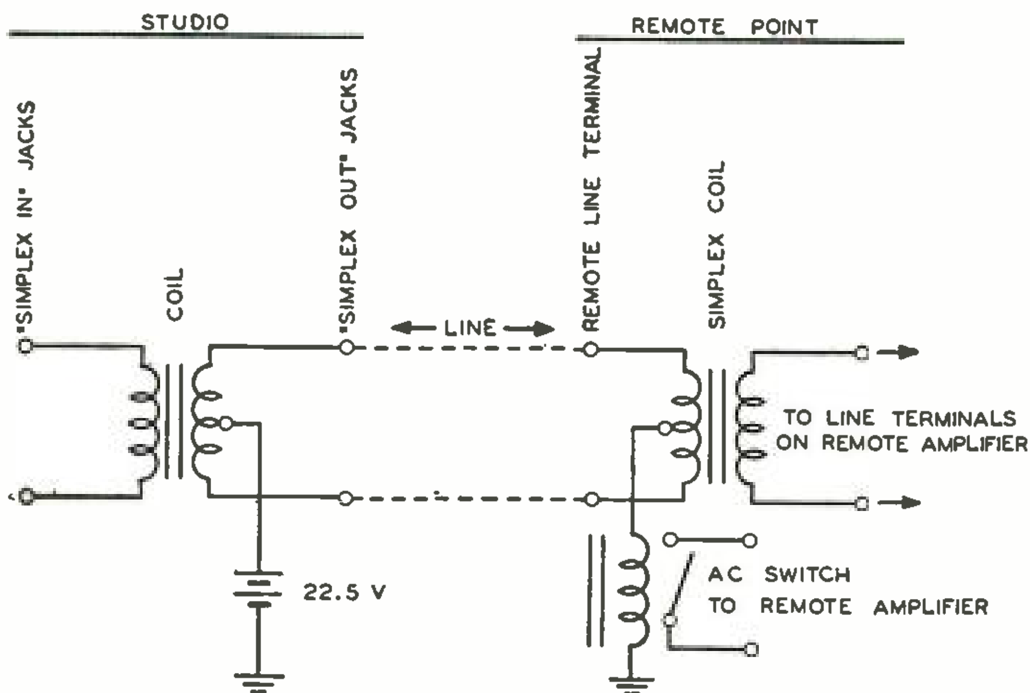


Fig. 11-2. A simplified schematic of a "simplex" installation which is used to switch a remote amplifier on and off from the studio.

is desirable for a regularly scheduled broadcast from a point requiring only one microphone where no "mixing" adjustments are necessary.

This may be accomplished by means of a "simplex" installation as shown in Fig. 11-2. The simplex coil "in" and "out" appears on the jack panel at the studio. When the remote line is patched into the equipment through this coil, it may be seen that the battery circuit will be completed to activate the relay at the remote point which is in the power-supply circuit of the remote amplifier. This procedure eliminates the necessity of sending an operator to this point each day.

OUTPUT CIRCUITS AND LINE EQUALIZATION

The line amplifier at the studio is always "isolated" from the line by a pad and a repeater coil. The necessity for this becomes apparent if the reader visualizes what would occur if such a means of coupling was not employed. An amplifier with an output of 600 ohms to match a 600-ohm line, if connected directly to the line, would have its frequency response materially affected by the length and characteristics of the line itself. Due to the distributed inductance and capacitance of the line, a different impedance would exist at every different frequency. For this reason, a pad of at least 10-db attenuation is always used to load the output of the amplifier, followed by what is known as a "repeating" coil. The control-room equipment is nearly always "balanced to ground" to prevent hum pickup and cross talk, in which case the repeat coil has a center tap to ground on both primary and secondary. When the equipment is single ended (one side grounded), the repeat coil is single ended on the primary side and balanced on the secondary or line side in order to provide a suitable connection of unbalanced to balanced conditions.

Operators and technicians are frequently confused when looking at the schematic diagram of the control-room installation to find a 600-ohm to 150-ohm output pad which is intended to feed a 600-ohm line. This arrangement is often used, however, where the line to be fed is comparatively short and unequalized. It should be remembered that the capacitive effect along the line attenuates the higher frequencies. Using a mismatch of this kind provides a beneficial equalizing effect which tends to compensate for the characteristics of the line. This arrangement is also often used on lines that *are* equalized, the amount of equalization necessary being less, with less insertion loss due to a great amount of equalization.

Line Equalization

Although the telephone company usually equalizes the incoming network lines and regular broadcast lines from studio to transmitter, many times it is beneficial to equalize lines from remote pickup points where the cost of high-class line service is not practical to the station. For this purpose, most stations have an equalizer of adjustable characteristics in the control room which may be used for this purpose.

Fig. 11-3 illustrates a typical setup for equalizing a broadcast line. The signal source is a steady tone from an audio oscillator which is terminated in an isolation pad. The same load at any frequency must be presented to the volume indicator and this instrument should, therefore, be bridged on the oscillator side of the pad. The equalizer must be on the line side of the coil at the receiving point, as shown in Fig. 11-3.

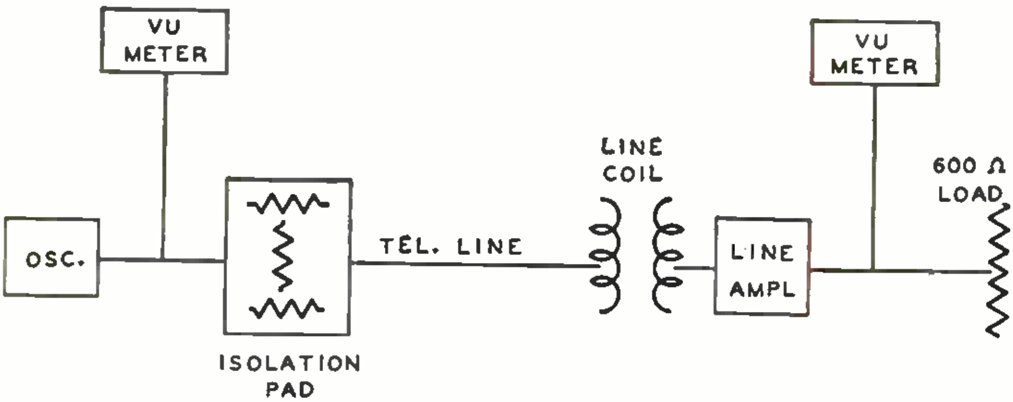
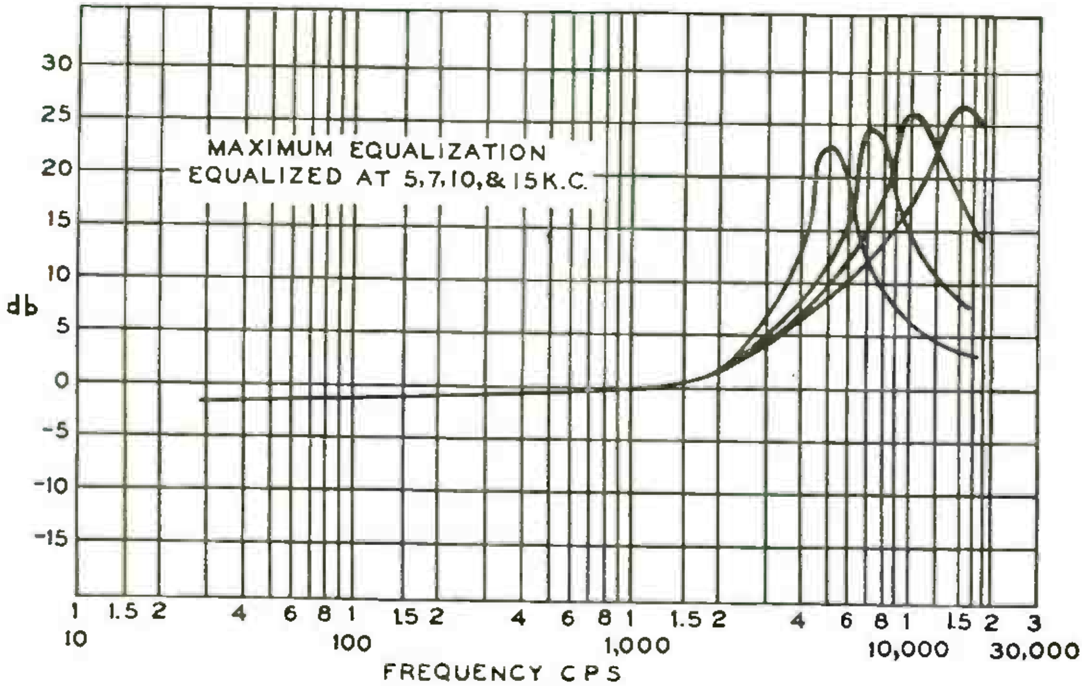


Fig. 11-3. Block diagram for a typical setup for equalizing a broadcast line.

A 1,000-cps tone is usually used as the reference frequency. The oscillator is set at this frequency and fed to the line at 0-vu level. The gain of the receiving amplifier is adjusted to give 0-vu reading at that point. The oscillator is then adjusted to 100 cps, 1,000, 3,000, and 5,000 cps with constant level maintained at each frequency, and the equalizer adjustment made at the receiving point to compensate as much as possible for the line characteristics at each frequency. This will determine the approximate setting of the equalizer, after which finer adjustments may be made over the entire frequency range.

Fig. 11-4 illustrates the shape of a typical equalizer curve, showing the maximum boost at 5, 7, 10, or 15 kc. Ordinary a-m services usually equalize to 5 or 7 kc, while f-m lines may use equalization to 15 kc. The insertion loss of this equalizer is approximately equal to the amount of equalization. A calibrated attenuator selects the



Courtesy Collins Radio Co.

Fig. 11-4. A typical equalizer curve showing the maximum boost at 5, 7, 10, or 15 kc.

amount of equalization at the required frequency which is selected by a panel switch. The input and output impedances are 600 ohms.

While considering the use of an equalizer, the reader should remember that some loss of signal energy will take place, and some form of extra amplification is necessary. A one- or two-stage isolation amplifier is usually used in conjunction with an equalizer to compensate for this loss, and should have at least a 30-db gain.

TELEPHONE COMPANY LINE SERVICES

Line services offered by Bell Telephone and American Telephone and Telegraph (AT&T) are divided into two general categories:

Metropolitan Area Circuits (Bell Telephone) which provide service for remote pickup points and studio-to-transmitter loops.

Toll Circuits (AT&T) composing the national network of circuits and long lines outside the metropolitan area.

The metropolitan area circuits are divided into services of the following general limitations:

1. Frequency range of 35 to 8,000 cps within plus or minus 1 db of 1,000-cps reference, and a volume range of 40 db. This service is sometimes used for studio-to-transmitter program loops.

2. Frequency range of 100 to 5,000 cps within plus or minus 2 db and a volume range of 40 db. This is the more common studio-to-transmitter service, and sometimes used for remote pickup points.

3. Nonloaded and unequalized commercial telephone service for use of remote pickup points.

Toll circuits are divided into general classifications such as high-quality, medium-quality, and speech-only services as follows:

1. Frequency range of 100 to 5,000 cps and volume range of 30 db. This is the normal service of national network hookups.

2. 150 to 3,700 cps, sometimes used where not enough time was available to install the higher-quality service.

3. Speech-only service of about 250 to 2,750 cps. Also often used for intercommunication between long-distance points and emergencies.

Line Isolation Coils

Line-to-line coils, usually termed "repeat" coils or "isolation" coils, have numerous uses in the field department of a broadcast station. They are almost a necessity in most instances when using older type or modified remote-control amplifiers having an unbalanced output circuit (one side grounded). Also some dubious designs of remote cue-back facilities using an unbalanced circuit will not permit the use of a public-address installation at the remote point without the use of such a coil. This will be clarified in the following discussion.

No electrical circuit can be called an "isolated" circuit, since moving electrons (which is what constitutes current flow) cause an electromagnetic field to be existent about the conductors. In addition to this fact, all conductors have some capacitive effect existent to some point, since, by definition, a capacitor is two conductors separated by a nonconductor.

Consider now the electrical characteristics of the telephone line used to carry broadcast program currents from studio to studio, studio to transmitter, and remote points to studio. Observation of Fig. 11-5(A) reveals the electrical nature of a line which has one side grounded. Electromagnetic voltages such as might be set up in the line by adjacent power lines may be represented by the generator shown in series with the line. Also, since some capacitance exists between the line and ground, any interference picked up by the capacitive coupling may be represented by the generator shown from line to ground. Program currents are shown by solid arrows, induced noise currents by the dotted arrows. It is obvious here that the two currents add together, and will be transferred to the transmitter equipment input.

Such lines are no longer used for broadcast services. Many times, however, the operator is faced with adapting a spare public-address, monitor, or recording amplifier to remote broadcast service. Such amplifiers are almost always of unbalanced output circuit design, which means that one side of the output is connected to the chassis or ground circuit of the amplifier. It is also often necessary for the broadcast operator to feed a supplementary public-address amplifier at the remote point from the output of his remote amplifier. Quite often this public-address amplifier will use an unbalanced input circuit which, if connected directly to the output of the remote amplifier, will ground one side of the circuit. Either of the above conditions will result in effectively grounding one side of the program line, with the results shown in Fig. 11-5(A).

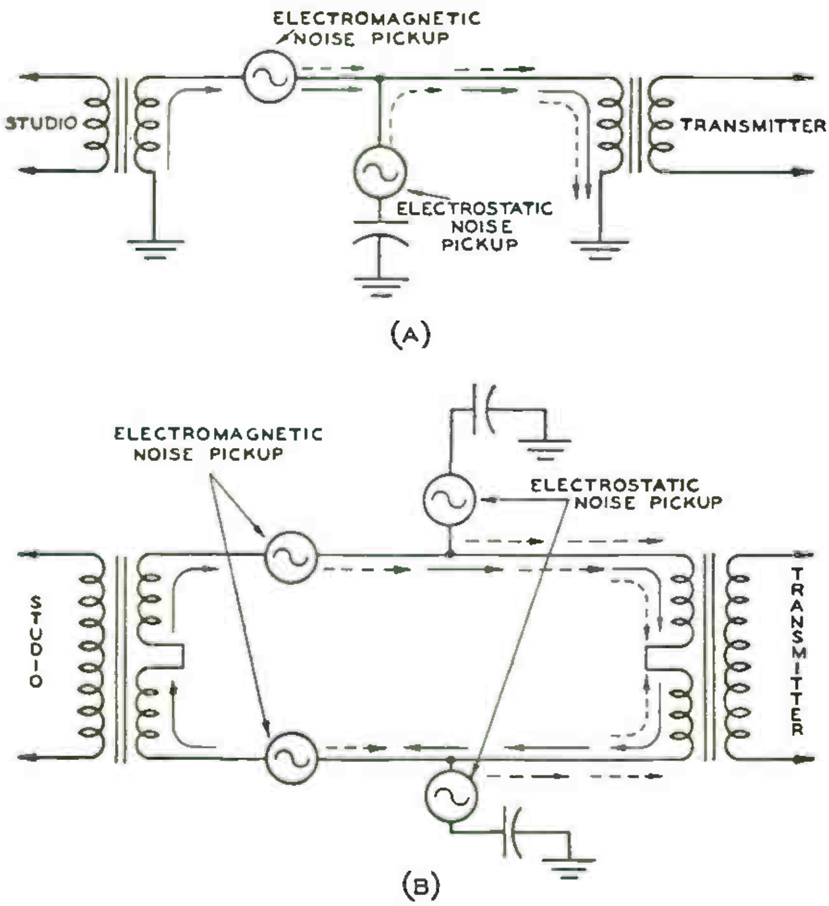


Fig. 11-5. Early types of telephone lines were grounded on one side resulting in noise pickup (A). Use of isolation coil eliminates noise pickup (B).

Telephone lines leased for broadcast services are always strictly metallic circuits (no ground return) as shown in Fig. 11-5(B), which shows the electrical characteristics of the line when balanced transformer windings are used at the terminating ends. It may be observed here that the noise currents from electromagnetic induction and electrostatic coupling are sent along the two wires in the same direction. Their respective directions of current flow, then, in the balanced transformer winding at the load end, will be 180° out of phase, and will cancel out. This leaves only the program currents which were induced in the line by the transformer at the sending end to be transferred to the transmitter input equipment. Some circuits ground the center tap of the windings, the majority, however, leave the center tap ungrounded.

Whenever it is necessary to use an unbalanced output circuit to feed a program line, an isolation coil should be used as shown in Fig. 11-6. This coil may be either 1 to 1 ratio (600-600) or a 600-150 coil may be used if the line to be fed is a long one. Such a mismatch as this provides a beneficial equalizing effect to compensate for the usual high-frequency attenuation of the line (see "Output Circuits and Line Equalization" earlier in this chapter).

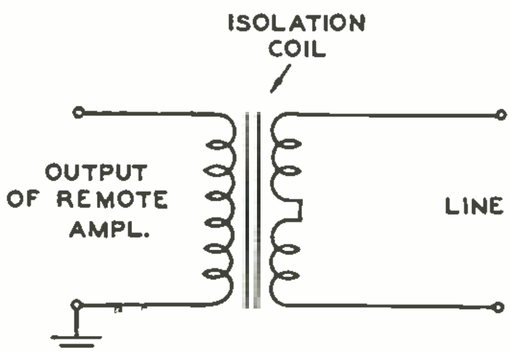


Fig. 11-6. An isolation coil should be used with an unbalanced output circuit.

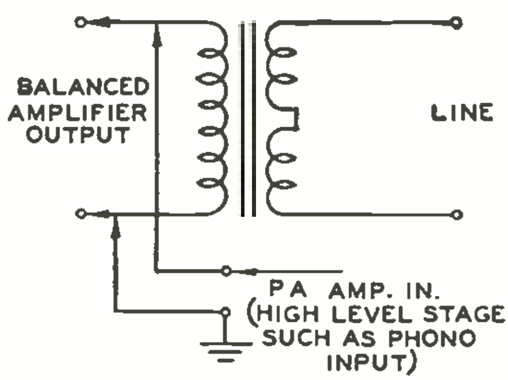


Fig. 11-7. An isolation coil should be used with an unbalanced input circuit.

At remote points where it is necessary to feed a public-address amplifier by direct connection, and the input circuit is unbalanced, the same type of coil should be used for the line connection. (See Fig. 11-7.) This permits the grounding of one side of the remote amplifier without disturbing the balanced line conditions.

As mentioned earlier, some remote cue-back circuits at the studio end use an unbalanced cueing circuit. Such circuits are not normally important since it is used only as a cue device. Consider, however, a case where it is necessary to feed the public-address amplifier this cue-back from the studio, which often occurs during certain types of programs where it is important for the audience at the remote point to hear preceding and succeeding signals from the studio.

Obviously, the unbalanced line signals will not be of a quality suitable to feed the PA system due to hum and noise on the line. Here again, it is necessary to use the coil as shown in Fig. 11-7, both at the sending and receiving ends of the line, so that balanced line conditions prevail.

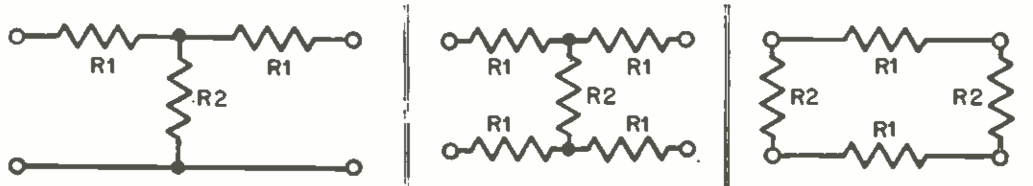
Program Line Levels

The program signal magnitudes on broadcast lines must be high enough for efficiently overriding noise level, but must be limited to prevent cross-interference into other lines by electromagnetic and electrostatic coupling. The maximum level permitted in most states is +12 vu into the sending end of the line.

The broadcast engineer faced with adapting spare amplifiers for remote amplifier purposes must keep this maximum level in mind. Public-address, monitoring, or recording amplifiers will generally have an output far in excess of allowable broadcast line levels. For convenience, Table 11-I with its associated values is presented here as a guidance in constructing suitable output circuit attenuating pads. The T pad is used for amplifiers with unbalanced output, in which case the isolating coil must be used as discussed above. For balanced outputs, the H pad or the O pad may be used, the H pad being preferred. It will be noted that the resistance values are seldom of standard values, being the absolutely correct values of resistance as figured theoretically. Substitution of the nearest value of resistance obtainable in RMA values will result in very little deviation of db loss value or impedance match. The values given are for 600-ohm circuits, which are standard for broadcast lines.

TABLE 1

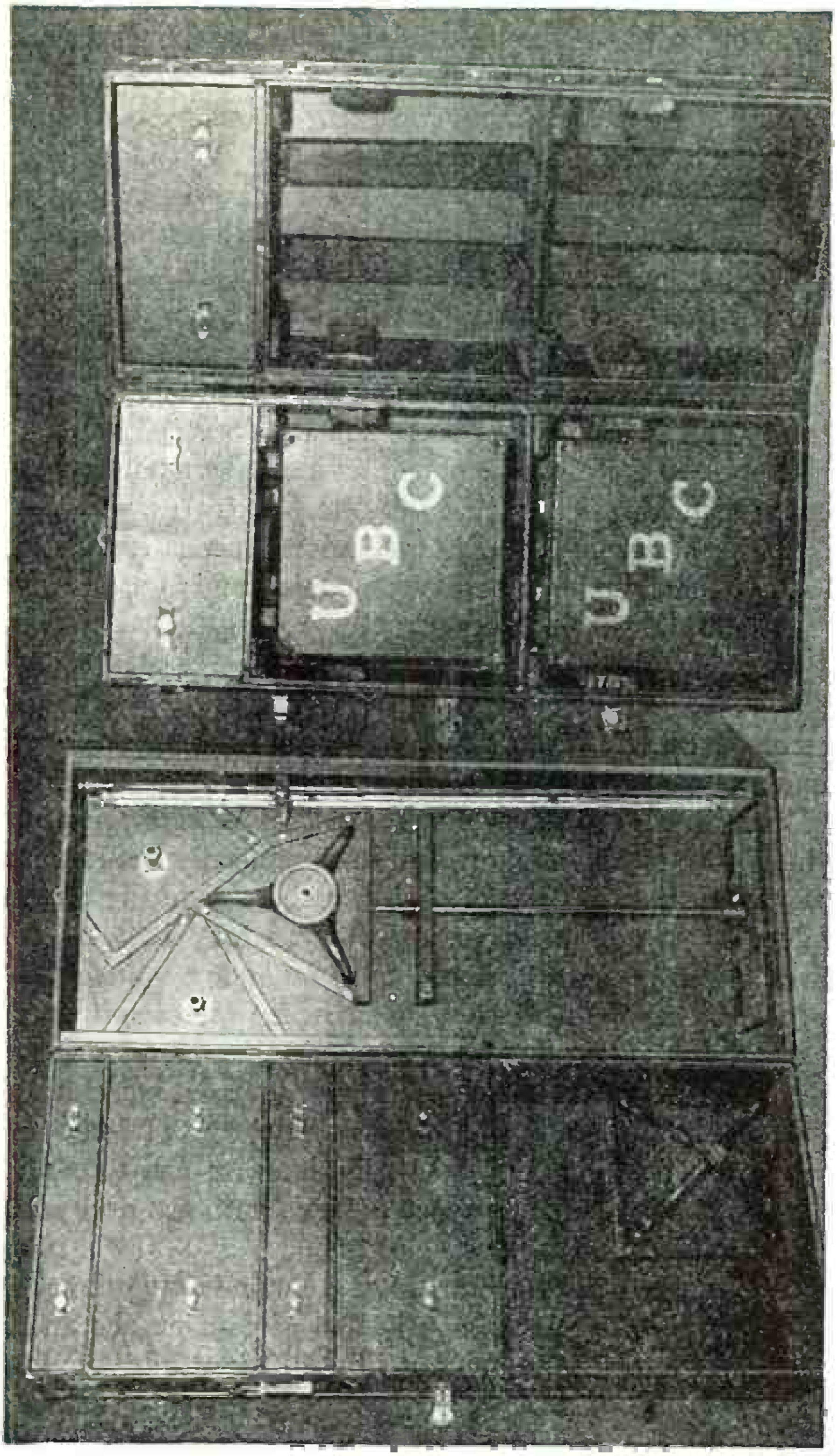
FIXED ALTERNATING PADS FOR 600-OHM CIRCUITS



	T Pad		H Pad		O Pad	
Loss in db	R1	R2	R1	R2	R1	R2
1	34.5	5208	17.25	5208	34.3	10440
2	68.8	2582	34.4	2582	69.7	5232
3	102.7	1703	51.3	1703	106.2	3505
4	135.8	1249	67.9	1249	143.8	2651
5	168.1	987.6	84.1	987.6	182.3	2141
6	199.3	803.4	99.7	803.4	223.8	1807
7	229.7	685.2	114.8	685.2	268.5	1569
8	258.4	567.6	129.2	567.6	317.1	1393
9	285.8	487.2	142.9	487.2	369.4	1260
10	312	421.6	156	421.6	427	1154
12	359.1	321.7	179.5	321.7	559.5	1002
14	400.4	249.4	200.2	249.4	721.5	899.1
16	435.8	195.1	217.9	195.1	923.2	826
18	465.8	152.5	232.9	152.5	1172	772.8
20	490.4	121.2	245.2	121.2	1485	733.3
22	511.7	95.9	255.9	95.9	1877	703.6
24	528.8	76	264.4	76	2369	680.8
26	542.7	60.3	271.4	60.3	2992	663.4
28	554.1	47.8	277	47.8	3775	649.7
30	563	37.99	281.6	37.99	4750	639.2
32	570.6	30.16	285.3	30.16	5967	630.9
34	576.5	23.95	288.3	23.95	7500	624.4
36	581.1	18.98	290.6	18.98	9480	619.3
38	585.1	15.11	292.5	15.11	11910	615.3
40	588.1	12	294.1	12	15000	612.1

Transporting Equipment

The field department is concerned with a very important task, the job of properly carrying delicate broadcast equipment outside the studio and reaching the remote point without the necessity of any major maintenance work to be done upon arriving. Microphones in particular must be firmly packed in such a manner that they do not roll around inside the case and must be protected against heavy physical shocks.



Courtesy WHK

Fig. 11-8. The larger trunk on the left accommodates the microphones, their stands, extension cords, tools, etc. The second trunk is designed for two remote amplifiers and battery boxes.

Field department engineers almost unanimously prefer microphone cords that are detachable at the head of the microphone. This allows extremely firm packing of the individual mike head between soft felt covers or rubber knee pads without worrying about excessive bends of the cord where it enters the mike housing.

Equipment transporting facilities sometimes take on an elaborate design for ease in handling a large amount of delicate equipment. Fig. 11-8 is a photo of the more complex design of broadcast equipment transportation trunk which was especially built by a trunkmaker. The larger trunk on the left has a number of drawers with sponge rubber lined compartments for microphones. The other half of this trunk is arranged for holding stand bases and shanks solidly as shown. The smaller trunk to the right accommodates amplifiers, battery boxes, long a-c and microphone extension cords, etc. Extra drawers in the trunk to the left accommodate tools and miscellaneous items. Such equipment must be handled by express companies, and is used only by larger stations and network centers.

Sparks and "Key Thump" Suppression

In some composite equipment not commercially designed, the engineer encounters "key clicks" or excessive sparking of relay or switch contacts. When such occurs, capacitor-resistor suppressor circuits placed in shunt with the contacts will aid in materially reducing such interference and increase the life of contacts where sparking occurs.

Exact values of capacitance-resistance to use may best be determined by trial. *The resistance should be of as high a value, and the capacitor of as low capacitance, as is effective for the circuit in question.*

Where the controlled current is pulsating or alternating, start with a capacitance of about 0.04 μf, and a series resistance of 1,000 ohms. If these values are not effective, gradually increase the capacitance and decrease the resistance for each capacitance until the desired suppression is achieved.

When sparking occurs across contacts of a noninductive direct current, the test runs should be started with 0.5 μf capacitance and 200 ohms resistance. The optimum values will not be beyond these limits.

Part 4

OPERATING THE TRANSMITTER

Chapter 12

OPERATOR'S DUTIES

ALL ENGINEERS and students familiar with the technical characteristics of transmitting equipment in general, and broadcast equipment in particular, are cognizant of the greatly advanced state of technical design and transmission fidelity. It will not be the purpose of this section to duplicate the already published data on broadcast transmitter circuit theory and relationships. A workable knowledge of this field is assumed in this text.

The discussion to follow will pertain to the all-important operation of the broadcast transmitting installation in order to achieve the best results possible from the finely engineered equipment available and in use today. Operating practice at the transmitter is just as important in the final result of over-all performance as it is at the broadcast studio. The science of operating the transmitter and associated speech input equipment may be shown to be a highly specialized art, and we have chosen the term "operational engineering" to define the content of the special study undertaken in this part of the handbook.

Outline of Responsibilities

It is true that the primary purpose of the transmitter operator is *to keep the station on the air*. But with the rapidly progressing demands for higher-fidelity program transmission, the day when the typical "ship operator" of thorough technical understanding could step into a broadcast installation, has passed forever. The operator of a broadcast transmitting plant has a specialized range of duties requiring a technical education, plus a thorough understanding and appreciation of the more intangible values of program material.

A number of his fundamental duties are, of course, strictly technical in nature and, since this is meant to be an analysis of an operator's duties, the technical functions will be described from an operational point of view. In brief, his technical duties consist in turning the transmitter on ahead of the beginning of the daily program schedule, checking all meter readings to make proper adjustments, checking level with the studio, shutting down the transmitter after sign-off, repairing and maintaining equipment, and testing for noise and distortion levels. During the daily operating schedule he consistently monitors the program from a monitoring amplifier and loudspeaker, adjusts line amplifier gain in accordance with good engineering practice pertaining to percentage modulation (the transmitter operator does not normally "ride gain" as does the studio operator), maintains correct tuning of transmitter, logs all meter readings every 30 minutes required by the FCC, and corrects any trouble that develops in the shortest possible time. Useful hints for meeting technical emergencies will be given later.

Typical Pre-Sign-on Procedures

The transmitter operator in all but the lowest power local stations is usually scheduled to be on duty at least 30 minutes prior to air time for the purpose of getting the equipment ready for the broadcast day. The start of an operator's day may be outlined as follows:

1. Audio rack power applied (including such measuring equipment as the frequency monitor and modulation monitor). Audio line used as program loop opened by inserting patch cord into the line jacks. This removes the line from the input to the line amplifier and prevents any test program that might be on the line from the studio from being applied to the transmitter when turned on.
2. Visual inspection of all relays in antenna-phasing cabinets (where used) and in coupling houses at the antennas. Relay armatures manually operated to ascertain freedom of movement. Observation of pointers on all r-f meters for bent hands or zero set.
3. Inspection of all safety gaps including antenna and transmission-line lightning gaps for approximate correct spacings.
4. Water pumps started (where used) and rate-of-flow meters observed for correct rate of water flow. Water flow must be normal before filament voltage is applied. Air-cooling systems usually start the blower motors when "filament-on" switches are operated. Transmitter filaments now turned on and filament voltages checked. In large power tubes using tungsten type filaments, minimum voltage should first be applied, then run up to normal filament voltage after about 3 or 5 minutes.

This is a means of lengthening the usable life of such power tubes but it is not observed in some stations. Tubes of the thoriated-tungsten or oxide-coated filaments such as used in the low-power stages, are always operated at *normal* filament voltage for maximum tube life.

5. Plate voltage then applied to low-power units or exciter unit (in power installations of 1 kw or more) to check for proper excitation to final stage.
6. Low power then applied to final stage. All meter readings checked for normal low-power operation. If everything is normal, high power applied and meter readings checked.
7. Filament and line voltages checked and adjusted for high-power operation. Final adjustment made on final stage for optimum meter readings regarding resonance condition and power input.
8. Since the control-room operator sometimes has circuits "hot" with his own testing procedure, the transmitter operator plugs patch cord from program line to monitor amplifier to ascertain continuity of program line. Then notifies control operator to stand by for over-all circuit test. When this has been done, transmitter operator removes patch cord from jacks which automatically restores line connection to input of line amplifier. A test tone may then be fed from studio to check over-all continuity of circuits from studio to transmitter modulators.

Pre-Program Level Checks

Level checks with the studio are not required as a daily procedure after an initial installation has been made, tested, and operated for some time, since with properly operating equipment the level remains very nearly the same over a period of time. At regular intervals, however, it is desirable to use a signal generator to check the frequency characteristics of the line and transmitting equipment. In this connection it is well for the transmitter operator to understand the difference in modulator power requirements for sine wave and speech or music program content.

It will be remembered from circuit theory that for a class C modulated amplifier, the power requirement for complete *sinusoidal* modulation is 50% of the d-c power input to the modulated tube or tubes. Fig. 12-1, however, shows how the "peak-factor" of speech or music waves varies greatly from that of a pure sine wave. This peak factor of program waves is 10 to 15 db *more* than that of a sine wave. That is, the ratio of peak to rms voltage is far greater for complex waveforms than that of a sine-wave form. In other words, the *average* power for complete modulation of a transmitter over a period of time is far less than the average power required for complete modulation by means of a signal generator. It is a well-known fact that for program

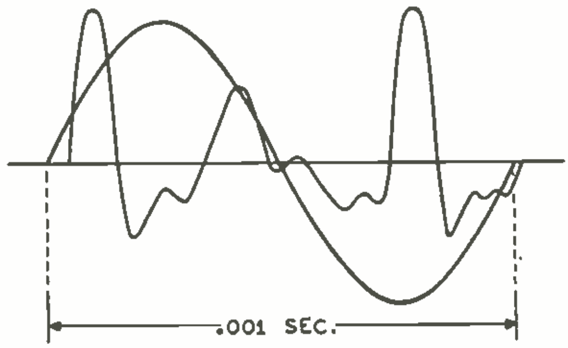


Fig. 12-1. The "peak factor" of a speech or music wave is 10 or 15 db greater than that of a sine wave; i.e. the ratio of peak to rms value is greater for complex waves than for sine waves.

signal waves the modulator power required may be 25% or less of the d-c power input to a class C stage. Therefore, if a signal generator is used at the studio for frequency runs or level checks, the transmitter operator must realize that if he has adjusted the gain on the line amplifier to give 100% modulation on sine wave, the same adjustment will be 10 to 15 db *high* for program signals. Thus the gain adjustment must be lowered to the point that experience has dictated for program modulation before the actual program schedule starts. In the past this has led to some confusion among transmitter operators.

This difference in peak factor between program and sine waves is also noticed when comparing the per cent of antenna-current increase with 100% modulation. It is true that the antenna-current increase should be approximately 22.5% over no modulation when a sine wave is applied to the transmitter at 100% modulation value. Antenna-current increases for 100% program modulation, however, will be much less, due not only to the difference in peak factor, but also the sluggishness of the thermocouple r-f meter action. This slowness of action is due to the heating effect of the two dissimilar metals upon which the action of the meter depends.

Frequency Monitoring

A frequency monitor must be included at all transmitter installations. This monitor is known as a "secondary standard," and is checked against a "primary standard frequency measuring service" at intervals of 30 to 90 days.

A station frequency monitor consists essentially of a crystal oscillator which operates at a frequency differing from the transmitter frequency by a given amount, usually 1,000 cps. This oscillator is beat with an r-f signal picked up from the transmitter and the amplified beat is applied across a diode circuit which drives the indicating instrument meter. (See Chapter 20 for a complete description of a typical frequency monitor.)

The standard broadcast station operator must ascertain that his transmitter is within ± 20 cps of the assigned frequency, the exact assigned frequency being indicated by zero (center position) on the meter. All oscillator units in transmitters have a small trimmer capacitor which enables exact adjustment of the frequency. As stated above, the monitoring unit itself must be checked against a primary standard at least every 90 days.

At f-m transmitters, the frequency and modulation monitor are always combined in one unit. (See Chapter 20.) The operator at a commercial f-m broadcast station must keep the center carrier frequency within ± 2000 cps of the assigned carrier frequency, indicated by zero (center position) on the meter.

Chapter 13

PROGRAMS *ARE* ENTERTAINMENT

DURING THE REGULAR broadcast day the transmitter operator keeps the circuits properly tuned, maintains correct power input to the final stage, logs meter readings each 30 minutes (which also aids in forestalling trouble), maintains frequency of operation within plus or minus 10 cycles, and maintains the program level at a point consistent with good engineering practice and the type of program in progress.

It is only natural that the program level being sent via wire line from the studio be the most concern from a strictly operational point of view to the transmitter operator. With competent studio personnel, the line amplifier gain adjustment may be set for 100% modulation on program peaks at the start of the day and left at that adjustment. Many times, however, the transmitter operator who does not appreciate musical and dramatic values will become piqued with the control operator when program level is very low. He should realize that broadcast stations are not strictly "communications," but intended to bring entertainment into the home with as much of the original intent as possible consistent with the state of the art. Certain types of programs, symphony concerts in particular, are meant for those listeners in the primary service area and not intended to override the noise level at some secondary service point. If the monitor speaker is turned up in volume consistent with that of the interested listener at home for these types of programs, the transmitter operator will be able to use good judgment as to whether the signal is entirely too low to be usable.

In relation to the study of program levels, it is of prime importance to understand the characteristics of indicating meters used at both ends of the transmission system. These meters differ in characteristics because of the different function which they are intended to perform. The standard vu meter used in most broadcast studios today is an rms-indicating full-wave rectifier device intended to give a close approximation visually of the sound waves emanating from the loudspeaker. We are concerned, however, with modulating voltages at the transmitter, and a semipeak indicating device is necessary and is required by the FCC. If peaks of the program signal content should be excessive and occur in rapid succession, danger of circuit component breakdowns would exist as well as severe adjacent channel interference. Therefore, since the peak factor of program waves is high as discussed earlier, the modulation meter is a "peak" indicating device. It is also necessary that a phase reverse switch be incorporated in the modulation meter circuit which switches the polarity of the input to the vacuum-tube voltmeter so that either the positive or negative side of the modulated envelope may be monitored separately. Thus it is obvious that we are confronted with two distinct types of level meters; namely, a *full-wave rms meter* at the studio and a *half-wave peak meter* at the transmitter. In addition to these meters, we usually have a limiting type amplifier (in most modern installations) which is used at the transmitter as a line amplifier. This has a meter which measures the amount of compression (full-wave peak meter) and output level in vu (full-wave rms meter).

Correlation of Meter Readings

The number of different types of indicating meters used should not confuse the operator as long as the proper interpretation is given to the readings. Fig. 13-1 is a representation of the indication of a pro-

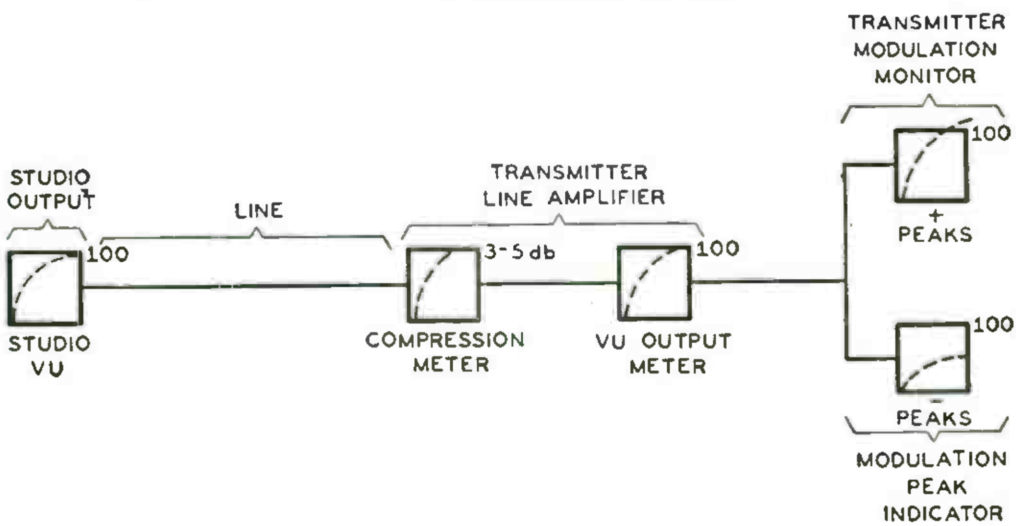


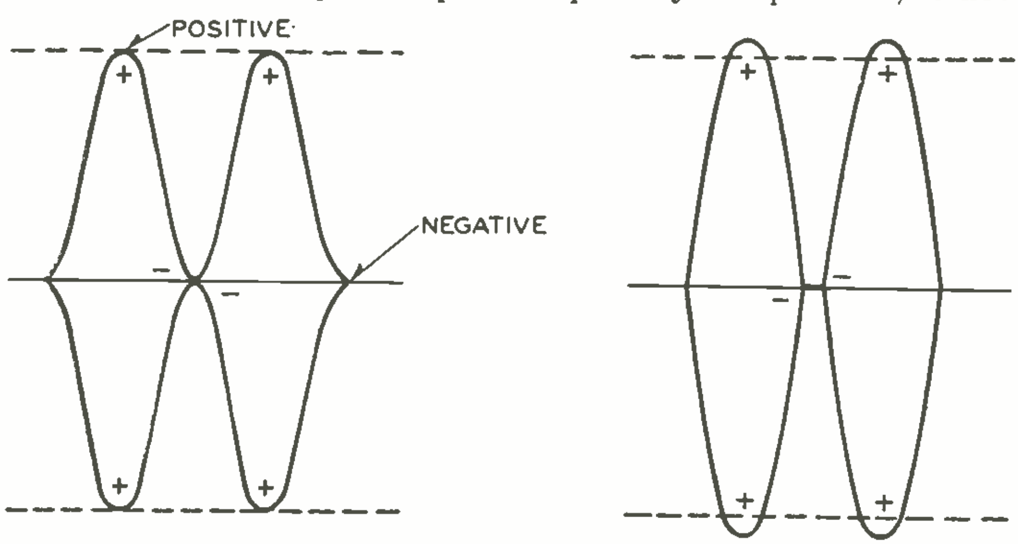
Fig. 13-1. A representation of the indications on the various meters of a program peak at any given instant.

gram peak *at a given instant* on the various meters involved. The studio vu meter has registered 100, the compression meter on the transmitter shows normal 5 db amount of limiting, the line amplifier output meter shows 100, and the modulation meter would show either 100% modulation on positive peaks, or, if set to monitor negative peaks, might show only 60% modulation. This, of course, could be just reversed with a change in polarity of the microphone output or any connection in between.

It is a well-known fact that speech waves are not equal in positive and negative peaks regardless of type of microphone used. This may be observed from the graph of the speech wave shown in Fig. 12-1. Two speakers working from opposite sides of a bidirectional microphone and "peaked" the same amount on the studio vu meter will *not* give equal indication at the modulation meter when set to indicate a certain peak (either positive or negative).

Assume, for example, that the modulation monitor switch is set to monitor the negative peaks, and the indication of one voice is close to 100%. The indication of the voice on the other side of the microphone (therefore oppositely poled at the microphone output transformer) may indicate only 40% or 50%, with the amplitude of the studio vu meter remaining the same. For this reason it is obvious why misunderstandings sometimes arise between studio and transmitter personnel regarding comparative level of two or more voices.

The question then arises as to what indication, if any, exists at the transmitter plant to show a true indication of comparative levels from the studio. It has been shown that the half-wave reading of the modulation meter, which depends upon the polarity of operation, is not a



Figs. 13-2, left, 13-3. A drawing (left) of an oscillogram showing a sine-wave carrier that is 100% modulated and one that is over-modulated is shown on the right.

true indication of *comparative* levels from the studios. The vu meter on the output of a limiting amplifier would not be a true indication since the output level is limited by the compression taking place in the amplifier for signals over a predetermined level. The compression meter, although a full-wave indicating device, is a peak reading instrument and, since the peak factor of program waves varies considerably, it is not an absolutely accurate indication of comparative levels. It is, however, the most reliable indication (within limits) existing at the transmitter, since it *is* full-wave rectified and is limited by only wire line characteristics. If two voices, for example, show about the same amount of compression, the comparative levels may be considered very nearly the same.

100% Modulation

Fig. 13-2 is a drawing of an oscillographic pattern of a 100% modulated (sine tone) carrier, showing what constitutes positive and negative modulation of the carrier. It may be seen that negative or "trough" modulation cannot attain more than 100% of the available range, whereas positive or "peak" modulation may go over 100%. When a carrier is thus modulated with a pure tone, the degree of modulation m is

$$m = \frac{\textit{average envelope amplitude} - \textit{minimum envelope amplitude}}{\textit{average envelope amplitude}}$$

and the peaks and troughs of the envelope will be equal. When the minimum envelope amplitude (negative peak modulation) is zero in the foregoing equation, the degree of modulation is 1.0 and the degree of modulation is complete, or 100% expressed in percentage modulation.

When the envelope variation is not sinusoidal, such as is true for program signals, the positive and negative peaks will not be equal as explained earlier, and the degree of modulation differs for peaks and troughs of modulation as follows:

$$\textit{Positive peak modulation} = \frac{E_{max} - E_{min}}{E_{av}} \times 100$$

$$\textit{Negative peak modulation} = \frac{E_{av} - E_{min}}{E_{av}} \times 100$$

Thus it is possible to understand the mathematical analysis of why the trough modulation cannot exceed 100%, since the minimum voltage cannot be less than zero. It may be seen, however, that the positive peak voltage may be more than twice the average (or carrier) voltage (E_{av}) in which case positive peak modulation will exceed 100% modulation. What important information does this hold for the transmitter operator?

First, it should be clarified in the operator's mind that "overmodulation" *can* take place on the negative (trough) modulation as well as on the positive (peak) modulation. It is true that the *degree* of modulation can never exceed unity on the negative peaks, but *can* exceed unity on the positive peaks. Complete modulation (of a class C stage) however, requires that the peak value of the modulating voltage equal the d-c plate voltage of the modulated stage. Fig. 13-3 shows a drawing of an oscillographic pattern of a carrier wave with modulating voltage exceeding the d-c plate voltage causing overmodulation of the carrier. It is true that the positive modulation peaks exceed unity while the negative peaks are "cut off" by the excessive negative modulating voltage and cannot exceed unity. This excess energy, however, which allows the negative peak voltage to result in a voltage applied to the r-f amplifier plate circuit to become negative with respect to ground, causes a radiation of this excess energy in the form of spurious frequencies, resulting in "splatter" and adjacent channel interference.

This actually is "overmodulation" in its severest form, since positive peaks may extend beyond 100% modulation without amplitude distortion, whereas negative peak "overmodulation" will cause severe amplitude distortion. It will be remembered that the bandwidth occupied by the carrier and sidebands depends (for amplitude modulation) *not* upon the degree of modulation, but upon the highest frequency being transmitted. Amplitude distortion, however, resulting from negative peak overmodulation, generates a number of distortion frequencies at harmonics that may well extend high enough to spread the sidebands into channels adjacent to the assigned frequency of the transmitter in question.

This discussion has been presented in order to show the transmitter operator that the negative side of the modulation is the most important peak to monitor on the modulation meter, and to hold under 100% at all times. It is well to remember that the modulation meter of the vacuum-tube voltmeter type will *not* be able to indicate over 100% (negative) on the meter because the peaks cannot attain more than this value as shown before. This is the reason why a cathode-ray oscilloscope is invaluable at a broadcast transmitter to show negative peak "overmodulation," since the negative peak "clipping" shows up as white lines across the center of the modulated pattern. When the usual vacuum-tube voltmeter type of modulation indicator is used, the flasher should be set to flash at 90% or 95% modulation so that when observing negative peaks, the warning is given when the peaks go up *to* 100% modulating value.

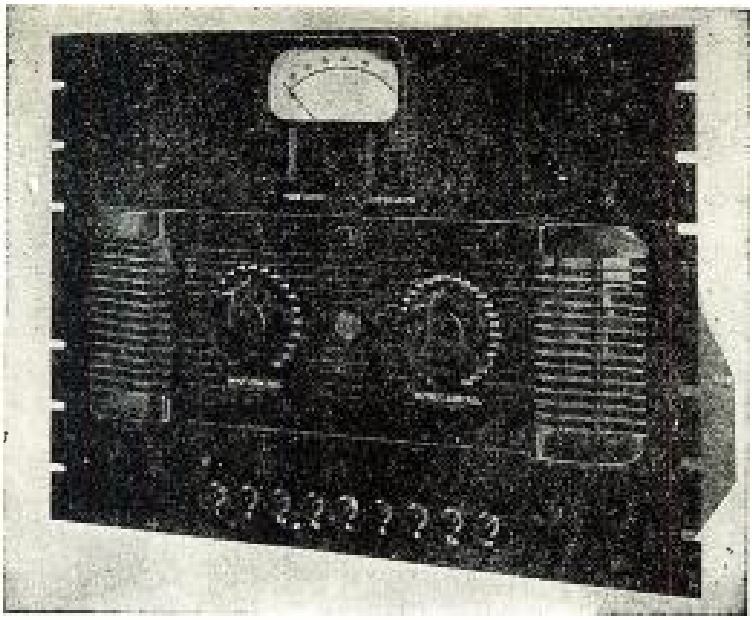
F-M Modulation Monitor Interpretations

Peak amplitude variations in the positive and negative directions show up on an f-m transmitter monitor as a decided difference in amplitude of the plus and minus excursions of the meter. In this case, however, we are not concerned with either peak in relation to distortion, since modern f-m transmitters are able to overmodulate greatly either plus or minus without inherent distortion. Distortion does occur, however, in the receiver when overmodulation over noticeable periods of time occurs, since few receivers will faithfully reproduce a frequency swing much over the maximum 150 kc ($\pm$75 kc) which is the value of 100% f-m modulation.

The f-m transmitter operator, therefore, is concerned with preventing overmodulation on either peak. For this reason, it is of utmost value to have all studio microphones poled so that their maximum polarity occurs on a definite side, either plus or minus on the modulation monitor. Otherwise, the transmitter operator must check his peaks at each change of microphone in use before he is certain that the maximum peaks are not over 100% value.

Operation of Limiter Amplifiers

The limiting amplifier, also known as a compression amplifier (see panel view in Fig. 13-4) is a very important link in a broadcast installation. However, its effect may be small and detrimental, if the wrong operational interpretation is given to the main purpose for which it is designed and intended. This type of amplifier, as designed for use in a broadcast installation, is intended as a *peak limiting* device, the amount of gain reduction being a function of the program *peak* amplitude. In order to prevent material reduction in the dynamic range of the signal, the peak gain reduction is not intended to be more than 3 to 5 db. A broadcast limiting amplifier, therefore, should not be considered as a volume limiter, but as a peak limiter intended to prevent overloading of transmitter components and adjacent channel interference.



Courtesy RCA

Fig. 13-4. Panel view of a limiting or compression amplifier for preventing overloading transmitter components and adjacent channel interference.

The original advertising claims of manufacturers offering this type of equipment proved misleading from an *operational* point of view. It is true that doubling the output power of a transmitter raises the signal intensity 3 db. It is also true that the limiter amplifier also raises the signal level about 3 db on *program peaks*. To those familiar with watching volume indicators on program circuits, however, this 3-db increase on speech or music is of small consequence. As far as the transmitter operator is concerned, he should think of this amplifier as a protective device to limit peaks caused by wire-line transmission and those program peaks that escape the action of the control-room operator.

That the primary purpose of a limiting amplifier may be defeated by erroneous operation is a very important fact for the broadcast operator to know. Seriously detrimental effects will result if this amplifier is operated as an actual "volume compression" device to attempt to prove a coverage area greater than a given power and transmitter location warrants. The "attack" time of peak limiting (about 0.001 second) is determined by a resistor-capacitor charging circuit with the inherent characteristics of a low pass filter. At high frequencies, and where the duration of the peak is short compared to this operating time, a portion of the peak energy will escape limiting action. If the average signal level is so high that a great amount of compression takes place at all times, a larger amount of adjacent-channel interference will result, thus defeating one of the main purposes of the amplifier.

This has been quite noticeable in practice when the program content consists of music from dance orchestras of brass instruments where high peak powers at high frequencies are very prevalent. A limiting amplifier operated properly for broadcast service will show about 3 to 5 db of intermittent gain reduction as indicated by the peak-reading meter used to show the amount of program peak compression. The operator must realize that, for certain types of programs such as symphonies, liturgical music, and operas, the average audio signal may be very low over a period of time even with limiting amplifiers in use. Dynamic range is just as important to high-fidelity transmission of these types of programs as is the frequency range.

Another consideration is the recovery time value, or time required to restore the gain to normal after a peak has momentarily reduced the gain. Optimum recovery time may well be different for different types of program material. Piano music, for example, sounds unnatural when recovery time is too short, because the effect is similar to inadequate damping of the strings after they are struck or to holding the sustaining pedal too long on the loud notes. The longer the recovery time is made, however, reduced gain will be, in effect, a larger proportion of the total time, and will result in unnatural transmission of certain passages in specific musical compositions. When operated properly in accordance with good operating practice, and not subjected to more than the specified amount of peak load, very satisfactory results may be obtained.[1]

When thinking of a compression amplifier as a means of increasing the service area of a transmitter, it is well to keep in mind the known facts concerning the psychological differences that exist in listening habits for various types of programs. A lower relative signal level is tolerable for dance music, news broadcasts, etc., where the average audio level is high over a period of time. In this case where listeners well outside the "primary service area" of the station may be numerous, the maximum amount of peak limiting may be used to help raise the signal-to-noise ratio at the receiving point. It is realized, however, that symphony broadcasts, choral music, certain liturgical

[1] Black, W. L., and Norman, N. C., "Program-operated level-governing amplifier," *Proc. I.R.E.*, Vol. 29, pp. 573-578, November, 1941.

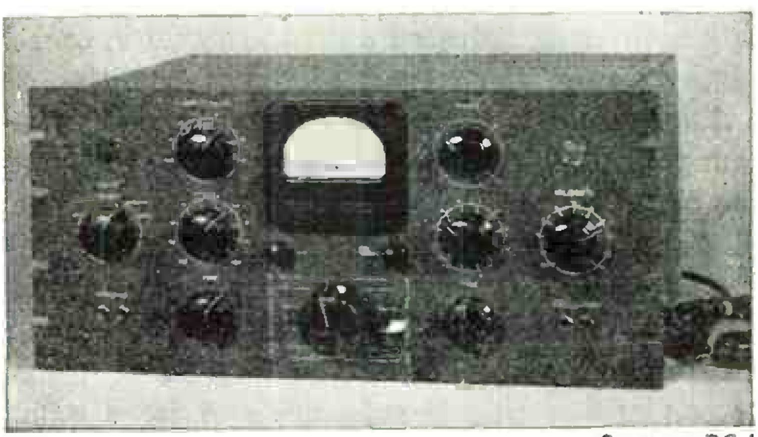
music, opera, etc., where the average audio signal may be very low over a period of time, will appeal only to those listeners who are very adequately served with strong carrier signals. In the interests of preserving the original dramatic effects of this type of program, *it simply is not technically feasible for a broadcaster to attempt to set a fixed value of coverage area for all types of program material.* Similarly, the operator responsible for the transmission of programs should not attempt to operate all equipment in the same manner regardless of type of programs being transmitted.

Chapter 14

MEASURING NOISE AND DISTORTION

THE IMPORTANT CHARACTERISTICS of any modern broadcast installation are adequate frequency range to convey as much of the original sound as possible, low noise and distortion levels necessary for required dynamic range, and dependability of performance. One of the most important pieces of auxiliary equipment about a transmitting plant is the instrument for determining noise and/or distortion over the usable frequency range to facilitate proper adjustment of the over-all installation. Several manufacturers are supplying such equipment for broadcast frequencies, and most stations are equipped with means of checking noise and distortion. Definite instructions accompany all such equipment, but a typical description of procedure for using one type of noise and distortion test equipment is given here in outline form as a matter of general interest. This outline conveys the general principles of all noise and distortion measuring equipment.

Fig. 14-1 is an illustration of the RCA 69-C Distortion and Noise meter, which may be used to measure distortion in transmitters or audio equipment at any frequency from 50 to 7500 cycles, providing



Courtesy RCA

Fig. 14-1. Meter for measuring distortion in transmitters or a-f amplifiers from 0.3% to 100% and noise levels down to —85 db below 12.5 milliwatts.

measurements of rms total distortion from 0.3% to 100% and noise levels down to minus 85 db below 12.5 milliwatts. This instrument consists essentially of a diode detector which is used when taking measurements on a transmitter to demodulate the modulated r-f signal,

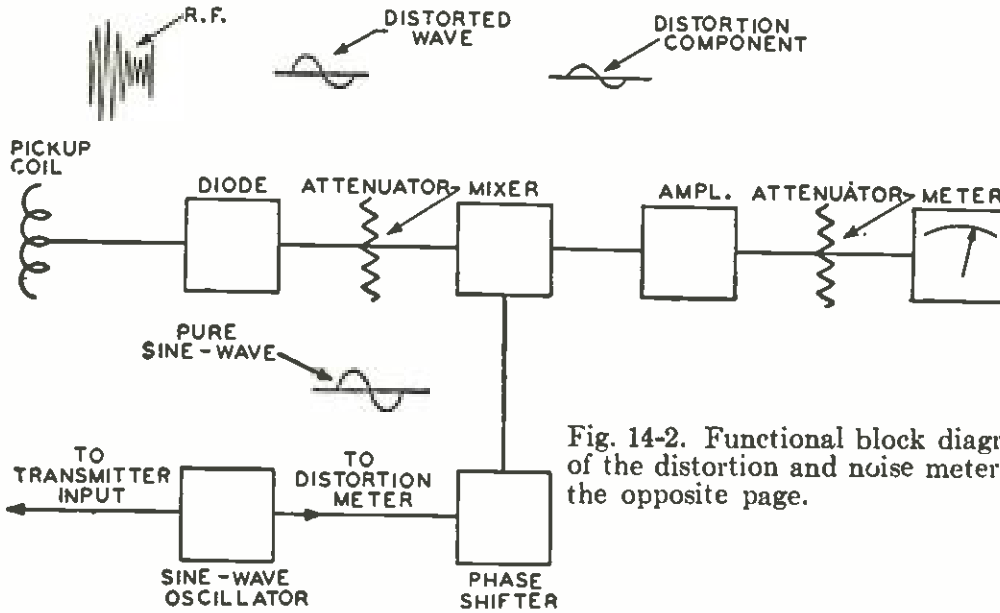


Fig. 14-2. Functional block diagram of the distortion and noise meter on the opposite page.

an attenuator to adjust the audio output of this detector, a phase-shifting network, a mixer stage to combine the output of the attenuator with that of the phase shifter, an amplifier, a range attenuator to adjust amplifier gain, and the meter which measures the amplifier output. A functional diagram is shown in Fig. 14-2.

In operation, a sine wave from a stable oscillator is applied both to the transmitter and the phase shifter of the RCA 69-C meter. The phase-shifting network is adjusted until the signal is exactly in phase with that derived from the output of the transmitter. The output signal from the transmitter is adjusted in amplitude by the attenuator so that its fundamental-frequency component is exactly equal to the output of the phase shifter. In other words, the amplitude and phase controls are adjusted until a minimum meter reading is obtained. Each of these two signals is impressed on the grid of one of two mixer tubes, whose plates are connected in push-pull by means of a transformer. The difference voltage of the two input signals appears across the secondary of this transformer and is at minimum value when the distorted and undistorted signals are adjusted in phase and amplitude to have minimum difference. With this adjustment, the fundamental-frequency component of the distorted signal is canceled out by the sine-wave signal, the difference voltage containing the distortion components. This difference voltage is amplified and the meter reads the total rms distortion directly.

Noise and distortion measurements should be run on broadcast transmitters at least every six months, as well as a complete frequency run to determine frequency response of the equipment.

RCA DISTORTION AND NOISE METER[1]

Description

The Type 69-C distortion and noise meter was developed to supply an accurate and reliable instrument for measuring the harmonic distortion and noise level in the output of radio transmitters, audio amplifiers, or modulated radio-frequency equipment of any type. Distortion or noise measurements are read directly from the meter scale, which is calibrated for several ranges. When used with the Type 68-A or 68-B low-distortion oscillator, distortion measurements may be made at any frequency from 50 to 8,500 cps or higher with weighting of the harmonics as indicated under Frequency Range. Reliable readings as low as 0.3 per cent may be made on any equipment having less than 180° phase shift throughout its frequency range. Under these conditions, the inherent distortion in the oscillator approximates 0.1% rms, which will have a negligible effect upon the distortion meter readings. Under the worst possible phase conditions, a residual reading of approximately 0.2% would be obtained.

Distortion measurements may be made at frequencies down to 20 cps with reasonable accuracy if the amount of distortion to be measured is not too small. Using 1 mw in a 600-ohm line as a zero reference level, distortion can be measured at volume levels as low as -17 db and noise levels may be measured as low as -75 db.

The essential elements of the 69-C distortion and noise meter are as follows:

1. An input circuit for the essentially sinusoidal signal from the 68-A or 68-B beat frequency oscillator, including a level control, marked "CALIBRATE," and a phase-shift network comprising three controls—coarse, medium, and fine, as shown in Fig. 14-3.

[1] Illustrations and other material furnished through the courtesy of RCA Mfg. Co.

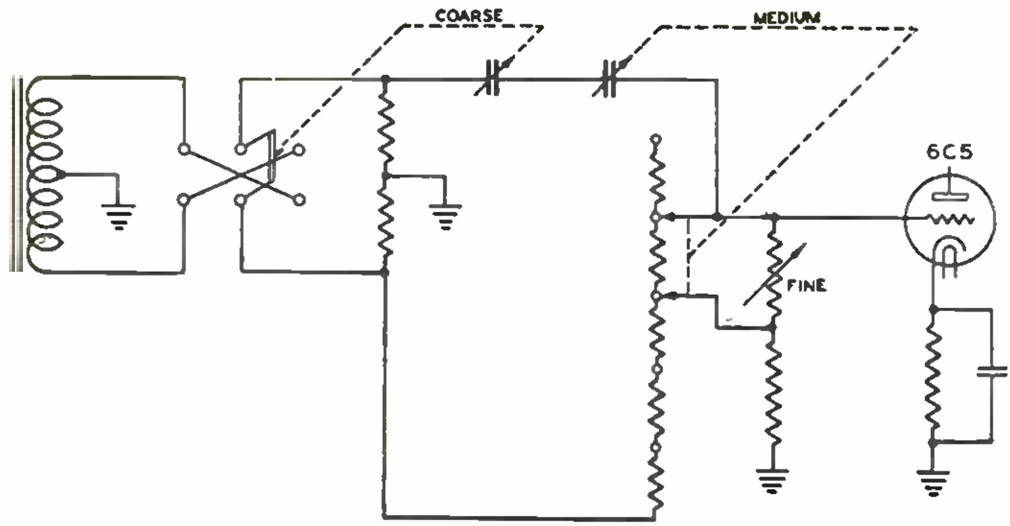


Fig. 14-3. Partial schematic of the distortion and noise meter showing the level control and the phase-shift network.

2. An input circuit for the distorted signal from the equipment under test. This includes a rectifier for demodulating an r-f signal when desired, a selector switch marked "INPUT," a source of voltage for standardizing the gain of a voltage amplifier, and three level controls —coarse, medium, and fine, as shown in Fig. 14-4.

3. A push-pull amplifier stage which is used as a normal amplifier for noise-level measurements, and as a cancellation stage for distortion measurements.

4. A "DISTORTION-NOISE LEVEL" switch, which is used for circuit switching and for controlling the attenuation between the push-pull amplifier stage and the voltage amplifier.

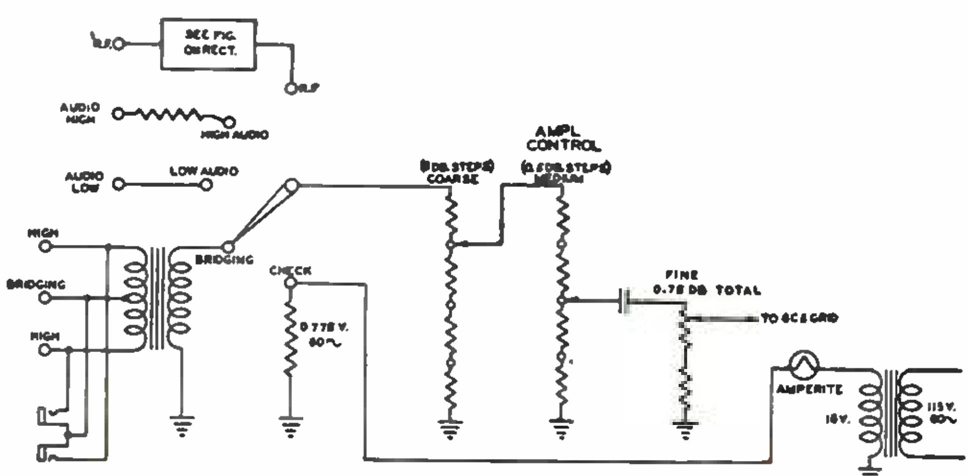


Fig. 14-4. Input circuit for the distorted signal from the equipment under test, including a rectifier, selector switch, and level controls.

5. A three-tube voltage amplifier with negative feedback. The "GAIN" control determines the gain of this amplifier by controlling the amount of feedback.

6. A detector and output meter, for measuring the rms value of signal. A small amount of bucking current is fed through this meter to buck out the no-signal plate current of the detector. The amount of bucking current is controlled by the "ZERO" control. Fig. 14-5 illustrates the distortion measurement circuit.

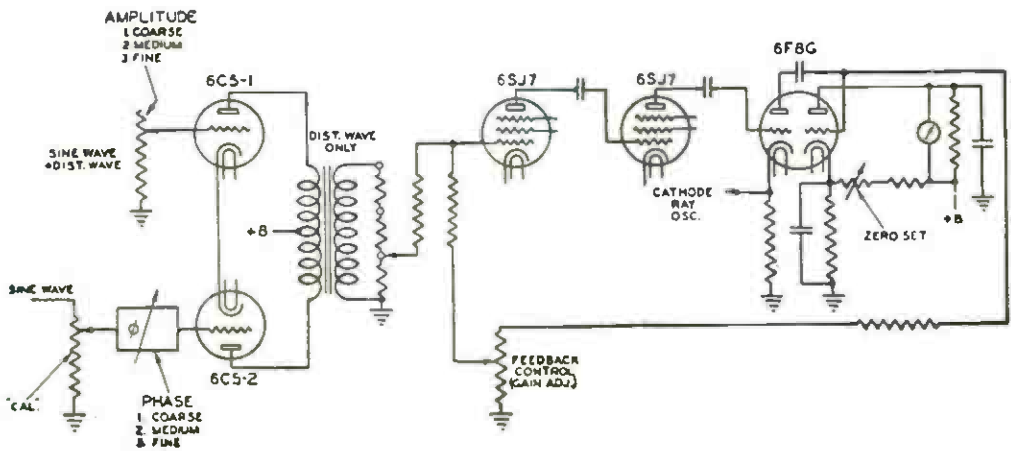


Fig. 14-5. Distortion measurement circuit consisting of a detector and output meter through which a small current is fed to buck out the no-signal plate current of the detector.

7. A power supply furnishing heater, plate, and screen voltages, and the standardizing voltage mentioned in 2. above.

In making distortion measurements, the meter indicates the distortion factor, that is, the ratio of rms total distortion to the fundamental amplitude. This is accomplished by suppressing the fundamental frequency component of the wave in question and measuring the rms total of the remaining components. Elimination of the fundamental frequency component is accomplished by adding to the distorted wave a sine wave of the same frequency, equal in amplitude to the fundamental component, but 180° displaced in phase. This voltage is secured from the same oscillator which supplies the signal to the equipment under test and is adjusted in amplitude and phase by the use of the controls on the panel of the distortion and noise meter. Distortion readings directly in per cent of the fundamental amplitude are obtained by first adjusting the meter to read full scale (100%) with only the sine-wave input connected.

Measurements of noise levels are made by adjusting the meter for full-scale deflection at the desired equipment output level and then removing the input signal from the equipment under test. The remaining noise and hum is amplified until a reliable meter deflection is obtained. The noise level is then read directly in decibels from the meter and attenuator scales.

Installation

The power cable should be connected between the a-c receptacle of the meter and a power-supply outlet furnishing 105-125 volts, 25-60 cps and delivering 50 watts. The power-line fuse on the chassis should be in the proper position corresponding to the applied line voltage. Terminals for connecting the distortion and noise meter with the associated equipment are located on the rear of the chassis with parallel-connected jacks located on the front panel.

The pickup circuit used for modulated r-f signals must provide a low-resistance d-c path between the r-f and ground terminals of the distortion meter as well as low audio-frequency impedance.

These conditions will be met by the use of a small pickup coil consisting of several turns. Capacitive coupling or an antenna may be used if a radio-frequency choke or a parallel resonant circuit is connected across the r-f and ground terminals. A low resistance, untuned coil is the most desirable for this purpose, as it is least likely to introduce hum into the circuit or to cause frequency discrimination.

The chassis of the distortion and noise meter should be well grounded to minimize stray r-f pickup. This can be accomplished by the use of a heavy strap or braid, as short as possible.

Operation

Distortion and noise measurements are read from the same meter, which is calibrated to the following full-scale readings:

Distortion	Noise Level
1%	—50
3%	—40
10%	—30
30%	—20
100%	—10
	0

The desired meter range is selected through the meter range switch, which is controlled by means of the large knob and scale. The desired distortion range may be selected by rotating the knobs over the left half of the scale. The desired noise-level range may be selected by rotating the knob over the right half of the scale.

INPUT LEVELS—For accurate distortion or noise measurements, the input levels to the instrument should be adjusted to within the following limits:

Modulated r-f – 10 volts to 80 volts.

To determine the proper r-f input level modulate the transmitter approximately 100% and set the "DISTORTION-NOISE LEVEL" switch at "0." Adjust the input level until full-scale meter reading is obtained with the "AMPLITUDE" control set between "0" and "+16."

Audio frequency from 68-A or 68-B oscillator—2 volts to 4 volts.

Audio frequency from equipment under test—

1. Bridging input terminals or jacks (balanced)
   - (a) Miminum—0.14 volts or –15 db below 1 mw on 600-ohm line.
   - (b) Maximum—9.0 volts or ±22 db above 1 mw on 600-ohm line.
2. Audio and ground input terminals
   - (a) "Audio Low"—0.12 volts to 8.0 volts.
   - (b) "Audio High"—1.2 volts to 80 volts.

COUPLING METHODS—Modulated radio-frequency voltages to be measured are obtained through inductive coupling. The pickup coil should be designed with a low audio-frequency impedance in order to eliminate any a-c hum component that may be picked up.

When the distortion and noise meter is to be used in conjunction with a balanced audio line having an impedance of 600 ohms or less, a bridging transformer having an impedance of 20,000 ohms is provided. This impedance is sufficiently high to have no appreciable effect upon the low-impedance line. The three transformer input connections terminate in three binding posts, marked "BRIDGING," located at the rear of the chassis, and a pair of parallel-connected jacks located on the front panel. The center tap of the transformer winding is not grounded.

CONNECTIONS—Following are tabulated the correct connections to be made for distortion and noise measurements under various conditions:

For Modulated Radio-Frequency Input—Connect the pickup coil between the "R-F" and "Ground" terminals at the rear of the instrument and remove all connections from the audio terminal. Set the "INPUT" switch to "R-F" position.

For Audio-Frequency Input Balanced Lines, Up to 600 Ohms—Connect the audio line either to the "BRIDGING" terminals at the rear or to the "BRIDGING" jacks on the front panel. The center tap connection may be connected, left open, or grounded as desired. Set the "INPUT" switch to "BRIDGING" position.

For Unbalanced Audio-Frequency Input—

1. Below 4 volts

Connect the audio line to "LOW AUDIO" and ground binding posts. Set the "INPUT" switch to "LOW AUDIO."

2. Above 4 volts

Connect the audio line to "HI. AUDIO" and ground binding posts. Set the "INPUT" switch to "HI. AUDIO."

For Distortion Measurements—Connect the 250- or 500-ohm 68-A or B oscillator terminals to the two terminals at the rear of the distortion meter marked "OSCILLATOR," or to the pair of jacks on the front panel marked "OSCILLATOR."

For Oscillograph Indication—When desired, a cathode-ray oscilloscope may be connected to the "CRO" binding posts to observe wave form of distortion or noise, or to assist in balancing out the fundamental. Any circuit connected across these binding posts should have an impedance of at least 100,000 ohms, and when an r-f field exists, such as around a transmitter, a shielded lead should be used.

CALIBRATION—Prior to making measurements, the instrument should be calibrated in the following manner:

1. Turn the power on by rotating the "CALIBRATE" control in a clockwise direction, and wait at least five minutes to allow voltages to stabilize.

2. With no input signal to the "OSC." binding posts or jacks and with the "DISTORTION-NOISE LEVEL" switch at the "CALIBRATE" position, adjust the "ZERO" control for a meter reading of zero per cent (not 0 db).

3. Set the coarse and medium "AMPLITUDE" controls to "0" positions and the "FINE" control with the pointer approximately vertical. Also set the "DISTORTION-NOISE LEVEL" switch to the "0" posi-

tion, and the "INPUT" switch to the "CHECK" position. Adjust the "GAIN" control for full-scale meter reading (0 db).

NOISE LEVEL MEASUREMENTS—Noise levels may be measured in either of two ways. One method gives a result in terms of the standard zero level of the 69-C, which is 1 mw in a 600-ohm line. The other method gives a result in decibels below some arbitrary output level of the equipment under test. The first method is accomplished as follows:

1. When using "LOW AUDIO" input, it is only necessary to remove input from equipment under test and adjust the "AMPLITUDE" controls and the "DISTORTION-NOISE LEVEL" switch until the meter reads on scale. The noise level (based on a 600-ohm line) is then read from the control settings and the meter readings.
2. When using "HI. AUDIO" input, a close approximation can be obtained by using the above procedure and adding −20 db to the result.
3. When using "BRIDGING" input, a close approximation can be obtained by using the above procedure and adding −1.5 db to the result.

The second method, which is the most accurate, is accomplished as follows:

1. Adjust the input to the device under test to obtain the output level below which it is desired to measure the noise level.
2. Adjust the "AMPLITUDE" and "DISTORTION-NOISE LEVEL" controls to obtain a meter reading of "0" db.
3. Remove the input signal from the device under test and move the "AMPLITUDE" and "DISTORTION-NOISE LEVEL" controls until the meter reads on the db scale. The sum of the amount that it was necessary to move the controls and the established meter reading denotes the noise level with respect to the original level.

DISTORTION MEASUREMENTS—Audio measurements—

1. Apply input signal from the low-distortion oscillator to the "OSCILLATOR" input of the distortion and noise meter, place the "DISTORTION-NOISE LEVEL" switch on "CAL." and adjust the "CALIBRATE" control for a full-scale meter reading. This setting should remain unchanged.

2. Adjust the input to the equipment under test to the desired level, remembering that output must be within the limits specified in input levels above.

3. Place the "DISTORTION-NOISE LEVEL" switch on "0," the "INPUT" switch on appropriate position, and adjust the "AMPLITUDE" controls for full-scale deflection of the meter.

4. Place the "DISTORTION-NOISE LEVEL" switch on "100" and adjust the "PHASE" controls until meter reading is below the calibrated portion of its scale. Turn the "DISTORTION-NOISE LEVEL" switch to "30" and by further adjustment of the "PHASE" and "AMPLITUDE" controls, obtain a minimum meter reading, turning the "DISTORTION-NOISE LEVEL" switch for increased sensitivity as required.

With the selector switch placed on "CAL," during distortion measurements, the meter reading may vary with the position of the "PHASE" controls. This is a normal characteristic resulting in an error of not more than 10 per cent on the "% DISTORTION" scale indication. In order to eliminate this error, place the selector switch on "CAL," after adjusting the phase controls for a balanced condition and readjust the "CALIBRATE" control for a full-scale meter indication. A slight readjustment of the "FINE" amplitude control will then be necessary for the final balance.

After obtaining an exact balance, the amount of total distortion is obtained by reading both the "meter" and "switch" scales. After a reading has been taken, the switch should be returned to the "CAL" position before making any adjustments to the equipment, in order to protect the meter.

CIRCUIT LOADING—The output of the Type 68-A beat frequency oscillator should terminate in the correct impedance in order to secure minimum distortion of the oscillator signal. The correct terminating impedance is indicated at each pair of output terminals. To illustrate, an impedance of 500 ohms should be connected between the two terminals marked 500, or an impedance of 250 ohms between each terminal marked 500 and the center tap terminal. The Type 89-A attenuator panel will provide proper impedance loading.

EFFECT OF NOISE ON DISTORTION MEASUREMENTS—The Type 69-C distortion and noise meter indicates the rms total of all components of the input signal which fall within the limits of the frequency range. The exception is the fundamental frequency component, which is canceled by the voltage taken directly from the oscillator. The reading of the meter will, therefore, include the following components:

Component	Frequencies for 1,000-cps modulation (60-cps power supply)
Harmonics	2,000, 3,000, 4,000 . . 20,000 etc.
Modulation cross products between hum and fundamental	$1{,}000 + 60 = 1{,}060$
	$1{,}000 - 60 = 940$
	$1{,}000 + 120 = 1{,}120$
	$1{,}000 - 120 = 880$
	$1{,}000 \pm 180 =$ etc.
Modulation cross products between hum and harmonics	$2{,}000 \pm 60 = 2{,}060$ and 1,940
	$2{,}000 \pm 120 = 2{,}120$ and 1,880
	$2{,}000 \pm$ etc.
	$3{,}000 \pm 60 = 3{,}060$ and 2,940
	$3{,}000 \pm 120 = 3{,}120$ and 2,880
	$3{,}000 \pm$ etc.
	$4{,}000 \pm$ etc.
Hum components	60, 120, 180, etc.
Noise components	All frequencies

The distortion and noise meter sums all these quantities and thus indicates, as per cent distortion, the ratio of the sum of all undesired components to the fundamental frequency component. If it is desired to determine the distortion due to the harmonic and cross-product components alone, either of two methods may be used.

One method is to operate the equipment under test at a high output level, which results in making the hum and noise components negligible compared to the other components. Another method is as follows:

1. Measure distortion in the normal manner at the desired output level.
2. Measure the noise level in decibels, using the same output level as a reference level.
3. Convert the reading in decibels to per cent; for example, -40 db = 1%, -60 db = 0.1%.
4. These values may then be substituted in the following equation:

$$H = \sqrt{D^2 - N^2}$$

where H = total harmonic and cross-section distortion in per cent
D = distortion per cent obtained as per (1).
N = noise (in per cent) obtained as per (2) and (3).

When making distortion measurements, it should be kept in mind that the noise level in the output of the beat-frequency oscillator approximates 50 db below 1 mw and is substantially independent of the actual oscillator output voltage. While the design of the distortion meter is such that the effects of noise and distortion present in the oscillator output tend to be canceled out, in most cases the cancellation will be more complete for the distortion than for the noise components.

Therefore, it is desirable to operate the oscillator at as high an output as practicable, thus improving the signal-to-noise ratio to the point where the noise output of the oscillator (expressed in per cent of signal) is small compared to the per cent distortion being measured. High oscillator output may not always be consistent with the input voltage requirements of the meter, but this difficulty can be overcome readily by the use of one or two attenuator pads or the 89-B attenuator panel.

When operating the noise and distortion meter at a point remote from the oscillator, the effects of noise and distortion in the line may be great enough to affect seriously the accuracy of the measurements. Hence this type of operation is not recommended.

Normally, when taking measurements near 0 or 180° phase shift, a balance cannot be obtained at frequencies which are transmitted through the equipment under test with phase shifts which fall within these narrow limits. This, however, can be overcome by inserting a capacitor in series with one of the two outside terminal connections (not the center tap) between the distortion meter and oscillator. The value of the capacitor and the choice of which connection to use is best decided by trial.

Maintenance

Service generally consists of replacing tubes which have become defective through usage. All tubes should be tested at regular intervals in a tube tester.

The distortion and noise meter is protected by a 1.5-ampere fuse. Should the clips holding this fuse become unduly heated through improper contact, the fuse will blow. Hence the holding contacts should be free from foreign matter and hold the fuse firmly in place.

Resistance elements through constant usage, sometimes become altered in value. This change, if sufficiently great, will affect operation in that portion of the circuit in which the resistance element is located.

Check tube socket voltages against the values in the table below. In event that the check on the tubes does not remove the cause of fault, disconnect the distortion and noise meter from its source of power. With an ohmmeter, check through the entire equipment for continuity.

If such procedure shows the circuit to be intact, then check each element therein with the ohmmeter and compare the resistance readings of the resistors against the corresponding resistor given in the schematic, Fig. 14-6.

In testing capacitors for open, short, and leaky circuits, it is necessary to remove one side of the capacitor under test from the circuit in which it is connected. The probes of the ohmmeter are then placed across the terminals of the capacitor under inspection and from the nature of the ohmmeter deflection, the condition of the capacitor can be ascertained readily. In the event that R55A, C8, C8A, or T2 require replacement, it will be necessary to readjust R55A so that the Low Audio response is flat within ±0.5 db from 30 cps to 40,000 cps, and down not more than 1 db at 45,000 cps. Potentiometer R55A is located underneath the chassis, on the shield.

Tube Socket Voltages

(120-volt line, fuse in 120-volt position)

All voltages except filament are d.c. to ground, measured with a 20,000-ohm-per-volt voltmeter.

Tube	E_f a-c	E_p	E_{sg}	E_k	$E_p\#_2$	$E_k\#_2$
6X5G R-F Diode	6.3					
6C5	6.3	120	—	3.7	—	—
6C5	6.3	120	—	3.5	—	—
6SJ7	6.3	152	112	3.8	—	—
6SJ7	6.3	152	112	3.8	—	—
6F8-G	6.0	105	—	3.6	250	11.5
6X5-G	6.3	d-c out = 357, a-c pl. to pl. 600 volts rms				
VR105/30		105				
VR150/30		255				
Amperite 6-8	12.0					

Noise-Level Measurements in F-M Installations

In "proof of performance" tests for f-m installations, the FCC requires both a-m and f-m noise-level measurements. (See "Excerpts

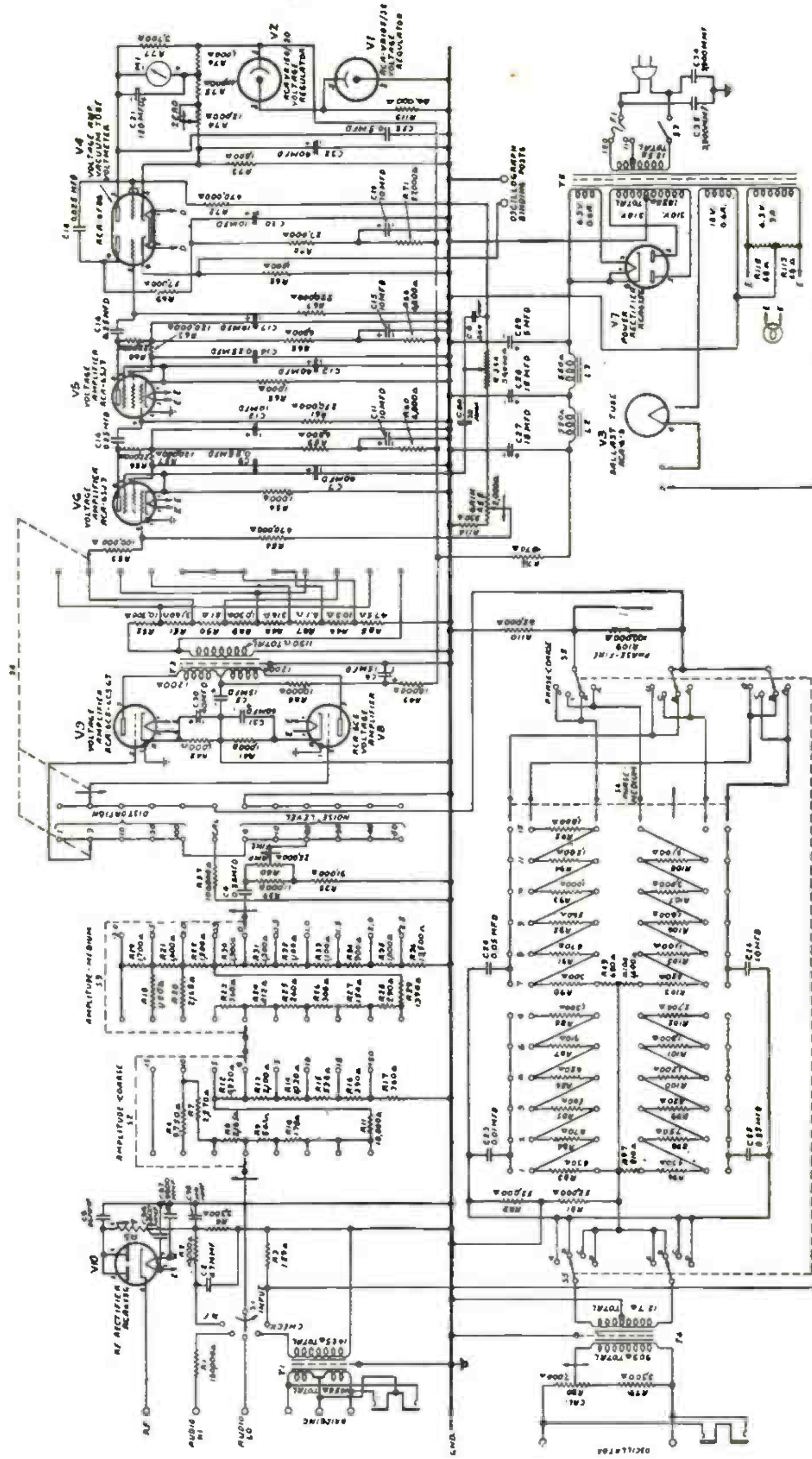
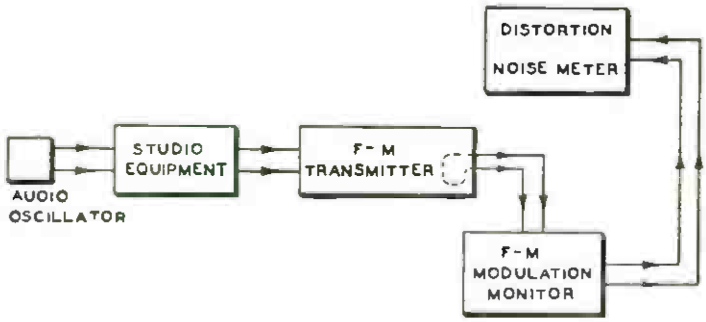


Fig. 14-6. Schematic diagram of the distortion and noise meter.

from Standards Relating to F-M Installations" later in this chapter.) Since this may cause confusion to the engineer concerned, an analysis of the two types of noise and their methods of measurement will be described herewith.

Fig. 14-7. Typical setup for making f-m noise-level measurements using a distortion and noise meter.



"F-m noise" is obviously a result of noise voltage components which *frequency modulate* the carrier wave. Thus any noise present in the program circuits, or any extraneous voltage induced in the audio circuits of the transmitter, will result in a frequency modulation and *f-m noise,* in addition to the desired program signal. Just as in a.m., f-m noise is measured in db below the level corresponding to 100% modulation.

"A-m noise" is any noise voltage component which causes *amplitude modulation* of an f-m carrier wave. In practice there is always some

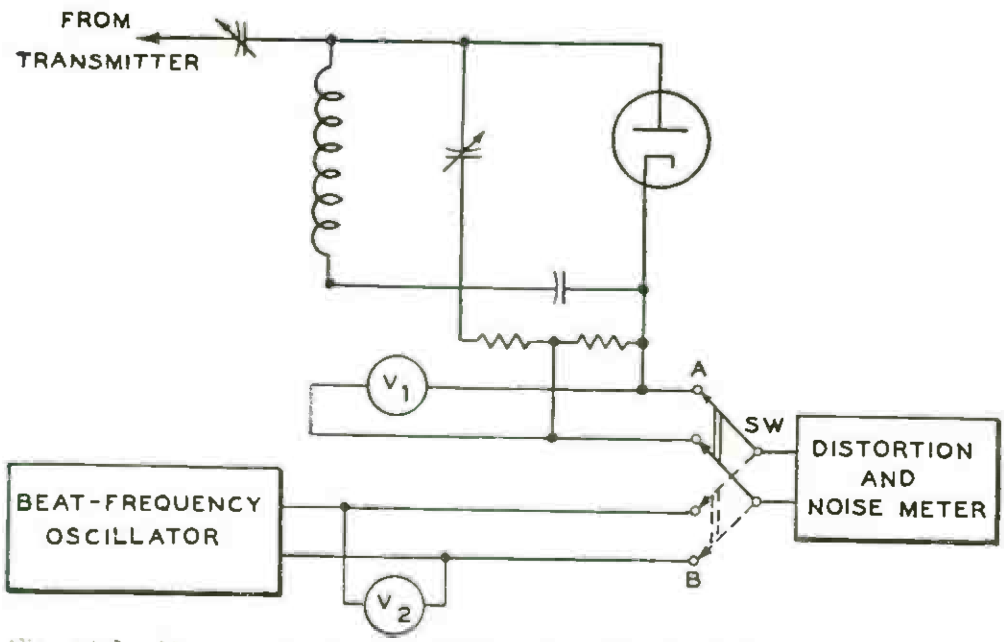


Fig. 14-8. Test setup for measuring a-m noise level in f-m transmitters using a distortion and noise level meter. The circuit shown here is used to obtain the a-m noise level that would correspond to 100 per cent a-m modulation of the transmitter.

slight amount of amplitude modulation in an f-m transmitter due principally to a-c operated filaments and extraneous pickup in the r-f circuits. Although theoretically such a.m. should not be passed on to the audio circuits of the receiver, such is not always the case in practice. Certain deficiencies of many f-m receivers, such as slight unbalance in the demodulator circuits, will result in a-m noise being passed on to the audio amplifier. The noise level, as in all preceding examples, is expressed in terms of decibels below what would correspond to *100% a-m modulation.*

The same equipment may be used in taking a-m and f-m noise-level measurements in f-m installations as used in the preceding examples of a-m transmitter measurements. A typical setup is shown in the block diagram of Fig. 14-7, for measuring f-m noise. Arrangement for measurement of a-m noise level in f-m transmitters is shown in Fig. 14-8.

Measurements of F-M Noise Level

Since the FCC requires an over-all level measurement for f-m noise, the setup of Fig. 14-7 is necessary. This includes, of course, the studio-transmitter line or r-f link where used.

In practice, the best demodulation system that can be used is the demodulator in the station's f-m modulation monitor. The monitor audio output terminals then may be connected directly to the audio input terminals of the noise meter. The steps of the procedure, as recommended by RCA, are as follows:

1. A 1,000-cps note from the oscillator is fed into the main studio speech input equipment (for example, into a microphone outlet) from whence it travels via the studio-transmitter link to the input of the f-m transmitter. The modulation monitor is left coupled to the transmitter in the usual manner, but the a-f output of the monitor is now fed to the noise meter, instead of the audio monitoring amplifier.

2. With the various gain controls set at approximately normal operating positions, the output of the oscillator is adjusted so that 100% modulation of the transmitter occurs as indicated by the modulation monitor.

3. Maintain 100% modulation, and adjust the noise meter for full-scale deflection. (This establishes a reference point.)

4. Turn off the oscillator and short the input terminals of the studio speech input equipment. The noise level in decibels below 100% modulation is now read from the noise meter.

Measurement of A-M Noise Level

Remember, in this case, it is necessary to obtain the a-m noise level in terms of what would correspond to 100% a-m modulation of the transmitter, and it is obviously not possible to amplitude modulate the f-m transmitter to 100%. Some other means must then be used to calibrate the noise meter.

The arrangement shown in Fig. 14-8 is recommended by RCA for the purpose. A diode rectifier with a 600-ohm output serves to rectify a small amount of r-f energy picked up from the output of the transmitter. A d-c voltage proportional to the carrier output of the transmitter appears across the 600-ohm output resistor of this rectifier. Should the carrier output be amplitude modulated, there would appear, in addition, an a-c voltage which would be proportional to the percentage modulation. If the carrier were amplitude modulated 100%, this a-c voltage would be equal to 0.707 times the d-c voltage. Thus we can set up an external calibration of the noise meter. This is done by using an audio oscillator of 600 ohms output so adjusted that there appears across the output an a-c voltage equal to 0.707 times the rectified d-c voltage measured above. This voltage is fed into the noise meter and the latter is adjusted for full-scale deflection. Thus calibrated, it is ready for use in measuring the a-m modulation level which appears across the output of the diode rectifier. The actual steps are as follows.

1. A diode rectifier as shown in Fig. 14-8 is coupled to the output of the transmitter. In some transmitters, (such as the RCA FM-1-B or FM-10-A) one half of the audio monitor coupling links may be used for this purpose.

2. With a switch *SW* in the *A* position, measure the d-c output voltage of the diode rectifier by means of the voltmeter V_1.

3. Throw switch *SW* to the *B* position and adjust the output of the audio oscillator so that voltmeter V_2 indicates an a-c voltage equal to 0.707 times the d-c voltage measured under 2. above. The noise meter is then adjusted for full-scale deflection. (Setting the reference.)

4. Return switch *SW* to the *A* position and read the noise level as indicated by the noise meter. This is the measurement of the a-m noise level.

Excerpts from A-M Standards

Section 3.46 requires that the transmitter proper and associated transmitting equipment of each broadcast station shall be designed, constructed, and operated in accordance with the "Standards of Good Engineering Practice" in addition to the specific requirements of the "Rules and Regulations of the Commission."

The specifications deemed necessary to meet the requirements of the "Rules and Regulations" and "Good Engineering Practice" with respect to design, construction, and operation of standard broadcast stations are set forth in the following text. These specifications will be changed from time to time as the state of the art and the need arises for modified or additional specifications.

A. Design. The general design of standard broadcast transmitting equipment [main studio microphone (including telephone lines, if used, as to performance only) to antenna output] shall be in accordance with the following specifications. For the points not specifically covered, the principles set out shall be followed.

The equipment shall be so designed that:

1. The maximum rated carrier power (determined by section 3.42) is in accordance with the requirements of section 3.41.

2. The equipment is capable of satisfactory operation at the authorized operating power or the proposed operating power with modulation of at least 85 to 95% with no more distortion than given in 3.

3. The total audio-frequency distortion from microphone terminals, including microphone amplifier, to antenna output does not exceed 5% harmonics (voltage measurements of arithmetical sum or rss) when modulated from 0 to 84%, and not over 7.5% harmonics (voltage measurements of arithmetical sum or rss) when modulating 85% to 95%. (Distortion shall be measured with modulating frequencies of 50, 100, 400, 1,000, 5,000, and 7,500 cps up to the tenth harmonic or 16,000 cps, or any intermediate frequency that readings on these frequencies indicate is desirable.)

4. The audio-frequency transmitting characteristics of the equipment from the microphone terminals (including microphone amplifier unless microphone frequency correction is included, in which event proper allowance shall be made accordingly) to the antenna output does not depart more than 2 db from that at 1,000 cps between 100 and 5,000 cps.

5. The carrier shift (current) at any percentage of modulation does not exceed 5%.

6. The carrier hum and extraneous noise (exclusive of microphone and studio noises) level (unweighted rss) is at least 50 db below 100%

modulation for the frequency band of 150 to 5,000 cps and at least 40 db down outside this range.

B. Operation. In addition to the specific requirements of the rules governing standard broadcast stations, the following operating requirements shall be observed:

1. The maximum percentage of modulation shall be maintained at as high a level as practicable without causing undue audio-frequency harmonics, which shall not be in excess of 10% when operating with 85% modulation.

2. Spurious emissions, including radio-frequency harmonics and audio-frequency harmonics, shall be maintained at as low a level as practicable at all times in accordance with good engineering practice.

3. In the event interference is caused to other stations by modulating frequencies in excess of 7,500 cps or spurious emissions, including radio-frequency harmonics and audio-frequency harmonics outside the band plus or minus 7,500 cps of the authorized carrier frequency, the licensee shall install equipment or make adjustments which limit the emissions to within this band or to such an extent above 7,500 cps as to reduce the interference to where it is no longer objectionable.

4. The operating power shall be maintained within the limits of 5% above and 10% below the authorized operating power and shall be maintained as near as practicable to the authorized operating power.

5. Licensees of broadcast stations employing directional antenna systems shall maintain the ratio of the currents in the elements of the array within 5% of that specified by the terms of the license or other instrument of authorization.

6. In case of excessive shift in operating frequency during warm-up periods, the crystal oscillator shall be operated continuously. The automatic-temperature-control circuits should be operated continuously under all circumstances.

Excerpts from F-M Standards

The transmitter shall operate satisfactorily in the operating power range with a frequency swing of ±75 kc, which is defined as 100% modulation.

The transmitting system shall be capable of transmitting a band of frequencies from 50 to 15,000 cps. Preemphasis shall be employed in accordance with the impedance-frequency characteristic of a series inductance-resistance network having a time constant of 75 microseconds. (See Fig. 14-9.) The deviation of the system response from the standard preemphasis curve shall lie between two limits as shown in Fig. 14-9. The upper of these limits shall be uniform (no deviation from the 75-microsecond time constant) from 50 to 15,000 cps. The lower limit shall be uniform from 100 to 7,500 cps, and is 3 db below the upper limit; from 100 to 50 cps the lower limit shall fall from the 3-db limit at a uniform rate of 1 db per octave (4 db at 50 cps, bottom of dotted line); from 7,500 to 15,000 cps the lower limit shall fall from the 3-db limit at a uniform rate of 2 db per octave (5 db at 15,000 cps, top of dotted line).

At any modulation frequency between 50 and 15,000 cps and at modulation percentages of 25, 50, and 100%, the combined audio-frequency harmonics measured in the output of the system shall not exceed the rms values given in the following table:

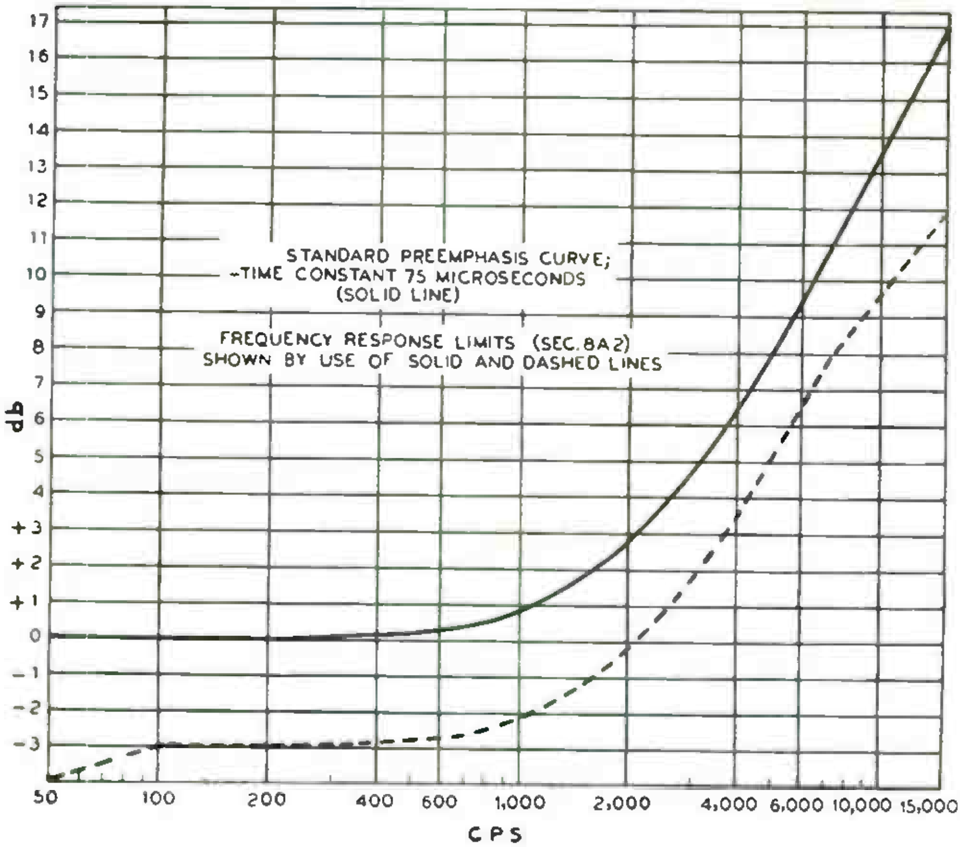
Modulating frequency	Distortion per cent
50 to 100 cps	3.5
100 to 7.500 cps	2.5
7,500 to 15,000 cps	3.0

Measurements shall be made employing 75-microsecond deemphasis in the measuring equipment and 75-microsecond preemphasis in the transmitting equipment, and without compression if a compression amplifier is employed. Harmonics shall be included to 30 kc.

It is recommended that none of the three main divisions of the system (transmitter, studio-to-transmitter circuit, and audio facilities) contribute over one-half of these percentages since at some frequencies the total distortion may become the arithmetic sum of the distortions of the divisions.

The transmitting system output noise level (frequency modulation) in the band of 50 to 15,000 cps shall be at least 60 db below the audio-frequency level representing a frequency swing of ± 75 kc. The noise-measuring equipment shall be provided with standard 75-microsecond deemphasis; the ballistic characteristics of the instrument shall be similar to those of the standard vu meter.

The transmitting system output noise level (amplitude modulation) in the band of 50 to 15,000 cps shall be at least 50 db below the level representing 100 per cent amplitude modulation. The noise-measuring equipment shall be provided with standard 75-microsecond deemphasis: the ballistic characteristics on the instrument shall be similar to those of the standard vu meter.



Courtesy FCC

Fig. 14-9. Standard FCC preemphasis curve for f-m transmitters. The area between the two curves is the deviation allowed for individual transmitters.

Spare Tubes. A spare tube of every type employed in the transmitter frequency and modulation monitors shall be kept on hand at the equipment location. When more than one tube of any type is employed, the following table determines the number of spares of that type required.

Number of each type employed:	Spares required
1 or 2	1
3 to 5	2
6 to 8	3
9 or more	4

An accurate circuit diagram and list of required spare tubes, as furnished by the manufacturer of the equipment, shall be retained at the transmitter location.

Operating Power: Determination and Maintenance

A. The operating power of f-m broadcast stations shall be determined by the indirect method. This is the product of plate voltage

E_p and plate current I_p of the last radio stage, and an efficiency factor, F; that is:

$$\text{Operating power} = E_p \times I_p \times F$$

The efficiency factor F shall be established by the transmitter manufacturer for each type of transmitter for which he requests FCC approval, and shall be shown in the instruction books supplied to the customer with each transmitter. In the case of composite equipment the factor F shall be furnished to the Commission by the applicant along with a statement of the basis used in determining such factor.

B. The operating power shall be maintained as near as practicable to the authorized operating power, and shall not exceed the limits of 5% above and 10% below the authorized power except in emergencies. In the event it becomes impossible to operate with the authorized power, the station may be operated with reduced power for a period of 10 days or less provided the Commission and the inspector in charge of the district in which the station is located shall be notified in writing immediately thereafter and also upon the resumption of normal operating power.

Part 5

WE'RE OFF THE AIR

Chapter 15

EMERGENCY SHUTDOWNS

THIS IS the situation that invariably causes a state of panic in the newcomer to a transmitter operating job. In nearly all instances he is alone, with the responsibility of correcting the trouble as quickly as possible to avoid loss of revenue by his employer. The highest efficiency in correcting trouble will come with more experience at the particular installation. The operator, however, who can visualize general circuit theory in relation to the particular circuits with which he is concerned will find a logical and natural sequence of looking for the fault. The *main* requirement quite naturally is to become thoroughly acquainted with the circuits used. He should be able to draw from memory a good general *functional* picture of all circuits, and be able to draw a block diagram of the sequence of operation of starting relays and protective relays in the power-control circuits. It is obvious that confidence and peace of mind can be achieved only by a complete familiarity with all circuits and their relation to the over-all performance of the transmitter.

It is, of course, impossible to set down a definite method of locating and clearing specific troubles of any kind or description. We hope, however, to be able to set forth a clear concise approach to procedures in general; that is, a logical and straightforward means of meeting emergencies.

There is one piece of equipment at the transmitter installation that should be the central focusing point for the operator's first attention when trouble occurs. This is the modulation meter which has an r-f input indication meter that reads a definite place on the scale for normal operation, and, of course, the percentage modulation indicator. The purpose of this will be evident in the following discussions.

At the first interruption of the program, or the occurrence of noise or distortion in the monitoring loudspeaker, this modulation monitor should be observed. Let us assume that the r-f input meter is at normal scale which assures us that the trouble is not in the r-f section because any trouble there would cause some deviation in the r-f input to the meter.

The following is a procedure to follow when the program suddenly stops from the loudspeaker:

1. *R-f input meter shows normal, modulation meter shows modulation taking place.* Trouble obviously in monitoring line or amplifier and we are *not* off the air.
2. *R-f input normal, no modulation as shown on meter.* Trouble either in audio section of transmitter, line amplifier, program line from studio to transmitter, or at studio. Call studio control to ascertain condition at that point. If everything there is normal, check line by patching line into monitor amplifier or spare amplifier to see if program is coming into the transmitter from line. If not, notify control to feed program on spare line and call local test board of Bell Telephone Co. If coming in satisfactorily from line, use spare line amplifier to feed transmitter. If the regular line amplifier is working normally, then the trouble obviously lies in the audio section of the transmitter itself. Usually any trouble here will be indicated by abnormal plate-current meter readings, and, of course, tube trouble is the most common source of program interruption.

The same procedure should be used where noise or distortion occurs, first checking with studio, then line, line amplifier, and audio section of transmitter. If all speech input tube currents are zero, then the trouble is in the associated power supply. Most likely trouble again is due to a tube, and it should be changed upon indication of abnormal plate current. Next in line comes bleeder resistors, resistor taps from bleeder supply, and line-to-plate circuits of tubes. Power-supply component parts usually show a visual indication of damage, such as a smoking part, unless opened up.

If, at the first indication of trouble, a glance at the modulation monitor position shows zero or low r-f input, then the trouble lies in the r-f stages of the transmitter. The operator must accordingly proceed to check for the trouble in the r-f section by observing all r-f circuit meter indications. Observation of plate and grid current meters aid in quickly determining the faulty stage.

When the transmitter is shut down by relay operation in the control circuits, the cause of the failure is quickly traced if the operator is familiar with relay sequence and functions. Control circuits are divided into two functional purposes: (1) those which control circuits to the primaries of power supplies, and (2) those of protective functions. Pilot lights are often associated with the various relays to show when they are open or closed. As stated before, the sequence of operation should be committed to memory. The filament power supply, for example, will not operate until the cooling motor contactors have functioned to supply the cooling medium (water or air) to the tubes. After the filament contactor has applied filament voltage, the plate-voltage contactor will not operate until the time delay relay has functioned, etc.

Rectifier tubes of the mercury-vapor type nearly always arc-back several times before expiring. When arc-back indicators are used, the faulty tube may be observed quickly and changed immediately. Other troubles in high-voltage power supplies nearly always show signs of physical deterioration as stated before.

Short circuits which cause a quick tripping of overload relays are always the most difficult troubles to locate. In some difficult troubles of this kind, overload relays have been strapped out of the circuits, and limiting resistors put in the lower current fuse box to limit the amount of current flowing. The circuits were then visually observed for arc-overs with doors open and interlock switches short-circuited. This is a dangerous procedure, however, and should be left to the more experienced operators. More than one man should carry out any unusual procedure of this kind.

This all may be summarized into the most important factor. *Be familiar with the transmitter,* and know what indications would be for the most common sources of trouble such as tubes and power supplies for the various circuits.

Be Mentally Prepared

We have discussed the art of being mentally prepared for emergencies in Chapter 5 on studio emergency technique. Obviously, the same sort of preparedness is an important factor at the transmitter for most efficient routine in mastering technical emergencies.

The first step is to get so familiar with the circuits proper that they may be drawn in whole or any part in simplified schematic form. This is only the first step, remember, in getting acquainted with the technical layout. The second step is to get just as familiar with the physical orientation of circuit components. Where is the main rectifier time delay relay? Where are the interlock contactors? Where is the modulator bias relay? Take the complete (not simplified) schematic "behind the doors" of your transmitter and visualize each component in the physical layout in relation to the schematic. In most modern commercial transmitters of the vertical chassis type construction, the schematic numbers are stamped either directly on the parts or on the chassis adjacent to the particular component. For example, if a certain coupling capacitor is shown on the schematic as *C102*, this number will appear on the capacitor itself or on the chassis adjacent to the location of the capacitor. In some cases only the tube numbers (*V1*, *V2*, etc.) and transformer numbers, (*T1*, *T2*, etc.), are shown. The parts associated with a particular tube may be oriented easily by tracing their connections to the proper tube socket terminal or transformer terminals.

When the operator feels at ease with these first two steps, he is ready for the third all important step, getting mentally prepared for trouble in his particular technical layout.

What specific example can we give as illustrating mental preparedness at transmitters? One very important example is this: will *reduced power* help to stay on the air, either until the end of the day's schedule, or until a sustaining program comes along to avoid loss of revenue? This will, of course, be determined by the nature of the trouble. Any capacitor, resistor, tube, transformer, insulator, or power leads that would undergo less strain by a power reduction, obviously might hold a while longer by this means. Also, in the case of tripping of overload relays in rapid succession when the source of the trouble is still not apparent, reducing power should be the very first step in attempting to stay on the air. This is advantageous, too, when help may be needed in cases of the more serious type where it is preferable to have extra help either in replacing a part or tracing troubles not recognized readily. The station may be kept on the air until help arrives to carry out such emergency provisions.

In this connection, we would like to pass on to the reader the "Emergency Procedure" rules at station WIRE for transmitter personnel. These were authored by Eugene E. Alden, chief engineer of the station and contain common sense routines applicable to any station.

Wire Transmitter Emergency Procedure

This sheet is posted in the operating room of the WIRE transmitter:

In Case of Failure

1. The first duty of the transmitter operator is to keep the station on the air, and, if it goes off the air, to return it as soon as possible. However, if you have a failure and are not able to get back on the air in a reasonable length of time, (within one or two minutes) you are to advise the studio operator of the condition. The studio operator will then stand by to give assistance, making telephone calls to get in touch with the chief engineer, or any other assistance the transmitter operator may request.

2. Operators not on duty, and listening at home, may call in to ascertain if they can be of any help, but they are not to call the transmitter. They should call the studio operator who will keep them advised.

TRANSMITTER OPERATING ROOM REPORT

DATE 7-15-1947

TIME 5:34 pm AM PM

PROGRAM Melody Billboard (Musical ET)

ANNOUNCER Fisher

CONTROL OPERATOR Busart ORIGINATED STU C (BT)

TRANSMITTER OPERATOR Ennes

REMARKS:

Carrier off thirty seconds due electrical storm.

Pigtail lead on directional antenna relay in phasing unit burned open. Replaced with temporary clip jumper.

BY H. Ennes

(FILE IN DUPLICATE WITH CHIEF ENGINEER)

Fig. 15-1. A typical transmitter operating-room report. The one above was used by the author at Station WIRE in Indianapolis, Indiana.

3. In no event should more than two minutes elapse before advising the studio of a failure. Advise them sooner if you feel that there might be a chance of not getting back on in two minutes.

4. On the first sign of a possible failure, the Auxiliary Transmitter filaments should be turned on and preliminary adjustments made so that it can go on the air if the 5-D fails.

Note. If trouble develops on the antenna side of the directional relays, where the antenna system is common to both transmitters, it will not help to change transmitters. It is quite possible, however, that components that will not hold on 5 kw *will* hold on 1 kw.

5. Make it an important habit to note the *time* on *any* interruption to program, whether transmitter failure or lack of program from the studio. This will serve also as a double check on studio or network failures.

6. Try to be mentally prepared to meet an emergency. Inspect relays and lightning gaps often during an electrical storm. In this way you may see part of the trouble developing and can replace them, or have temporary clip leads ready to put in the circuit.

7. In case of time lost due to failure at the transmitter, report to the control room on the phone, time program stopped and time it started again.

8. Fill out "Transmitter Operating Room Report" form.

(Author's note: See Fig. 15-1)

Lightning and the Transmitter Operator

Mental preparedness at the transmitter is rigorously tested during electrical storms. In spite of the many provisions in modern transmitter installations to help protect the equipment from heavy lightning surges, (ball lightning gaps, automatic carrier off relays, etc.) the great majority of time lost in *well-maintained* stations is due to lightning.

There seems to be no rhyme or reason to some troubles that develop during storms. Regardless of critically spaced lightning gaps at the bases of the tower and both ends of transmission lines, lightning has been known to open or short interlock circuits, power supplies, speech input equipment, etc. The "shortest path to ground" relative to the antenna circuit is definitely *not* a 100% rule for lightning.

It is true, however, that a majority of failures due to electrical storms occur in the antenna or directional phasing equipment. The trouble is usually recognized immediately by visible indication of blackened or smoking parts. When line or antenna current meters are damaged by lightning, the face is nearly always so black as to be unreadable. Even when switch-blade shunts are kept on meters, lightning charges are apt to damage a transmission-line current meter by a heavy current arc from coil to magnet to ground. Obviously, when this happens it is necessary to remove the leads from the meter to remove the short to ground if this short remains after the initial hit. Relays in the antenna circuits are another common source of failure when severe lightning surges occur. It is very important that every transmitter operator have various sizes of clip jumpers handy to strap around any such failure. If only the relay holding coil is opened up, the relay may be blocked shut by some means of a prop, or weight, depending upon the type of relay and the method of mounting.

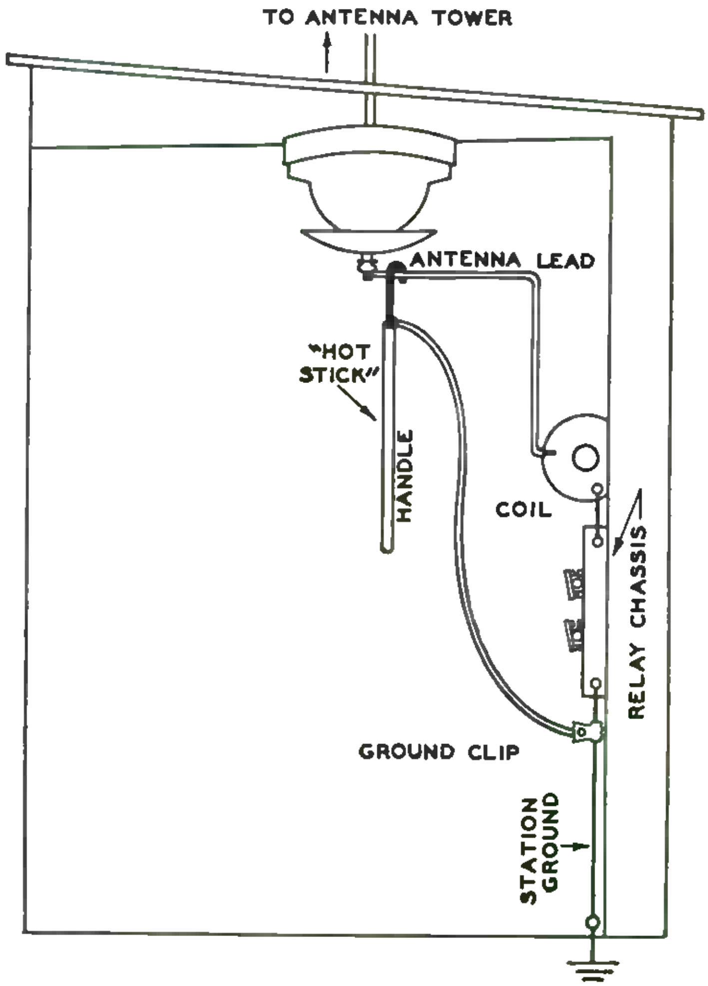


Fig. 15-2. The use of a "hot" or grounding stick is a necessary precaution when working on an antenna tower during an electrical storm or other similar disturbance. Hook up the hot stick as shown in this illustration.

Caution: Electrical storms are hazardous to a transmitter operator when work on the antenna system is necessary during the height of the storm. Never touch anything under these conditions until the tower has been well grounded by a "hot stick" or other arrangement. Remember to *always* clip the ground connection *first*, then hang the metal connection on to the tower lead as close to the actual tower base as possible. Fig. 15-2 illustrates such an arrangement. Grounding sticks of this type are an important and necessary item for any transmitter operating room.

Control Circuits

An operator cannot be employed over a period of several years without being faced with the situation of quick tripping of overload relays, finally resulting in a complete shut down of the transmitter. Modern transmitters employ an automatic return circuit so that three to five overloads must occur in rapid succession before the power is removed, requiring the notching relay to be reset by hand. Also, some form of visual indication is given as to which general section of the circuits where the overloads occur, such as D-C Power Amplifier Overload, D-C Modulator Overload, A-C Rectifier Line Overload, etc. The exact cause of the overload, however, is often hard to trace down, unless a visible arcing serves to indicate the source. Past history of the particular installation and observation of parts with voltage applied are helpful in most such instances. In some more serious cases, the cause is immediately obvious, such as a smoking capacitor or blackened high-voltage insulator with visible arcing under voltage.

Where arcing occurs, the hearing sense is usually able to locate the approximate vicinity where such is taking place. Upon opening the doors, signs of arcing should be visible such as burned spots on the frame immediately adjacent to a coil or capacitor corner.

The first emergency measure to take in any case of successive overloads that will not allow the carrier to hold, is to reduce power. If the carrier will hold on reduced power, the chief engineer or supervisory personnel may then be consulted to determine the best possible course of action from then on. When overloads must be traced down by strapping interlock circuits closed and opening the doors for visual observation with voltage applied, *more than one operator should be present.* Remember that no employer worth your services would require any other procedure.

Power amplifier overloads may, of course, be caused by trouble in the antenna system. Any fault that would cause grounding of the transmission-line center conductor or antenna (unless shunt fed) would cause the final stage to overload and operate the overload relay. Such faults may usually be found by visual observation. Look for blackened meter faces, smoking capacitors, or collapsed static drain choke causing a grounded tower to r.f. as well as d.c. Field mice have been known to get into antenna tuning houses and meet their end via direct r.f. to ground on the tuning component chassis. When examining any part of the transmitter or antenna system for possible sources of overloads, look around for such possibilities as mice, or bugs between plates of tuning capacitors, etc. The occurrence is more common than the newcomer might suspect.

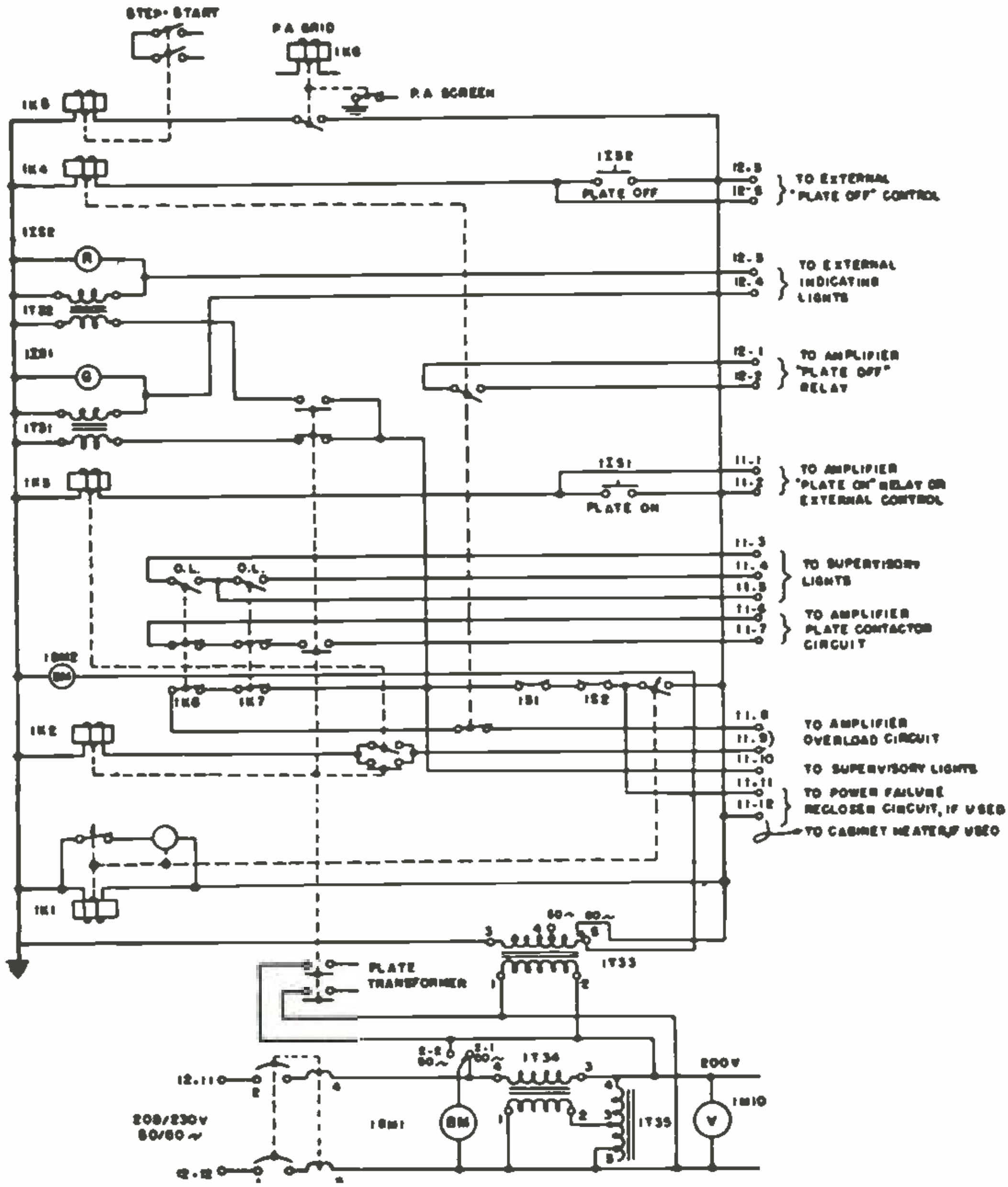
A-c rectifier line overloads that often trip circuit breakers or blow line fuses are almost always caused by a defective mercury-vapor rectifier tube which arcs back in inverse direction to normal conduction. Modern transmitters employ "arc-back indicators" which will give visual check on any tube in which current has passed in inverse direction. If tubes are visible from the front of the transmitter through the panel doors, a tube so afflicted will have its blue haze extinguished momentarily at the time of the arc-back, or a distinct flash will be apparent within the tube envelope. If rectifier tubes are not visible, the entire set of mercury-vapor tubes should be replaced with new ones. Remember that the new tubes must have been preheated (filaments lighted) for at least 30 minutes to assure that all mercury has been removed from the tube elements, and must not have been tipped over since this seasoning process. In some cases, the relay itself is at fault. If, for example, a holding coil should open up as sometimes happens, the simplest procedure is to prop or tie the relay shut manually. When this cannot be done conveniently, the proper terminal board numbers should be jumped to complete the circuit, or the jumper may be used at the relay contacts in some instances.

This type of emergency procedure may be made more obvious by going through a complete control system and studying the possibilities as follows.

F-M Transmitter Control Circuits

Fig. 15-3 shows the entire control-circuit arrangement of the General Electric 250-watt f-m transmitter type BT-1-B. There is no difference in the general application and theory of f-m control circuits and those used in a-m transmitters. Thus a thorough study of any particular circuit will provide a good background for all existing installations. The reader should refer to Fig. 15-3 constantly during the following explanation.

The entire control circuit operates from the secondary of *1T33*, energized when *1S3* is closed. When *1S3* is closed, the motor driven



Courtesy General Electric

Fig. 15-3. Control circuit arrangement for the General Electric type BT-1-B f-m transmitter.

timing relay *1K1* is energized and "times out" in approximately 30 seconds, which closes the normally open contact. A normally closed contact opens, removing power from the relay motor.

The green indicating light *1IS1* runs off of the transformer *1T31*. This transformer is energized now through the normally closed contact of *1K2*, provided rear door interlocks *1S1* and *1S2* are closed. This green light indicates that the transmitter is ready for application of plate voltage. The timing relay *1K1* has prevented application of plate voltage until tube filaments are warm.

A summary of what occurs upon closing switch *1S3* is as follows: The cabinet cooling blower *1BM1*, and tube seal cooling blower *1BM2* are energized. Also the control isolating transformer *1T33*, all filament transformers *1T6, 1T36, 1T37, 1T44, 1T45*, and *1T47*, variable transformer *1T35*, and bucking transformer *1T34*. These, of course, are in addition to the timing relay *1K1*.

Pressing momentary switch *1IS1* applies the plate voltage. This energizes relay *1K3*, closing the normally open contact. This closed contact now energizes *1K2* through the normally closed contact of *1K4* and the normally closed contacts of overload relays *1K7* and *1K8*. When *1K2* closes, power is removed from green light, *1IS1* and applied to red light *1IS2*. Also power is applied to the low- and high-voltage rectifier plate transformers *1T41* and *1T46*. Also a shunt is placed across the contact of *1K3* which locks *1K2* in, and *1K3* may be de-energized by release of *1IS1*. Also the plate contactor interlock between the 250-watt unit and an additional power amplifier (if used) closes *TB11-6* and *TB11-7*.

The momentary contact switch *1IS2* serves to remove plate voltage. Depressing this switch energizes *1K4* whose normally closed contact opens the de-energizing *1K2* which falls out, closing its normally open contact removing plate voltage from an additional amplifier (if used).

In series with *1K2* is one set of normally closed contacts on the low-voltage overload relay *1K7*, and one set of normally closed contacts on high-voltage overload relay *1K8*. Another set of contacts is extended for use in conjunction with a higher powered amplifier when used. Also a set of normally open contacts is extended to a supervisory light circuit when used. Thus an overload in the low- or high-voltage supply will cause *1K2* to drop out, will cause an amplifier plate contactor to drop out, and will cause a supervisory light circuit to operate and indicate in which circuit the overload occurred. These supervisory lights must be operated by auxiliary relays which will lock themselves in when the normally open contacts of *1K7* or *1K8* close for an instant. As soon as *1K2* falls out *1K7* or *1K8* will return to the energized position.

When relay *1K5* is energized (see below), the surge-limiting resistor *1R185* is shorted out. This resistor limits the peak charging current by the high-voltage rectifier tubes to a safe value when plate voltage is applied.

Relay *1K5* is energized through a contact of *1K6* after *1K6* is energized. Relay *1K6* closes when the power amplifier grid current exceeds approximately 8 ma. Previous to this a normally closed contact of *1K6* shorts the PA screens to ground, protecting the tubes from excessive dissipation before the grid excitation is sufficient to produce safe grid bias. By the time the PA grid current builds up to 8 ma and closes *1K6,* the high-voltage rectifier capacitors are almost fully charged. The subsequent energizing of *1K5,* and shorting of *1R185* then results in negligible surge current through the high-voltage rectifier tubes.

The principles of this control-circuit operation are the fundamental principles of all modern transmitter control circuits, and should become thoroughly familiar to all broadcast operators concerned with their use.

Control Circuit Emergencies

We are now able to study the possibilities of meeting emergencies that may arise from control circuit functions of this particular transmitter.

Suppose, for example, that filaments have been turned on by closing the main line switch *1S3.* Power is now also applied to timing relay *1K1,* which should light the green light after the customary 30 seconds of time delay.

This will occur *if:*

1. *1K1* is in working order.
2. The door interlocks *1S1* and *1S2* are closed.
3. The green indicating light *11S1* is in working order.

Now assume that after one minute the green light has not come on, although doors are closed (closing *1S1* and *1S2*) and should do so. We first examine *1K1* and find it has operated OK. How can we check the operation of the door interlocks *1S1* and *1S2* when the doors must be closed?

The most convenient procedure here is simply to jumper terminals *11-10* and *11-11,* (Fig. 15-3). It may be observed by the diagram that this completes the interlock circuit even though doors are open. It is then possible to go around to the front of the transmitter and observe the green light. If it is now on, one of the interlock relay coils is open or contacts are bad.

Suppose, however, that we had found *1K1* was not functioning, or that we suspected it. Observation of Fig. 15-3 shows us that a jumper from *11-11* to *11-12* would complete this circuit for a test. In case of extreme emergency where it is necessary to get on the air in a hurry and doubt exists as to which circuit is faulty, a jumper from *11-10* to *11-12* completes the entire circuit of timing-relay and door interlocks.

Go over the above procedure thoroughly until it is well understood. Remember, however, that it is an *Emergency* measure. Interlock circuits may be "strapped around" in this way when looking for arcs or malfunctioning equipment that cannot be seen with doors closed.

Understanding of the above type of emergency procedures will enable any operator to meet emergencies involving any of the circuits suspected of trouble.

See Chapter 17 for proper procedures of relay servicing.

To familiarize the newcomer with typical troubles at broadcast transmitters, the following case histories from an actual transmitter report file are presented. It may be observed that, in this particular installation, an auxiliary transmitter was available for emergency use. Note also how previous observations of certain symptoms aided in finally solving a particular trouble.

CASE HISTORIES

Oct. 16-1947. Time 5:32 P.M. Modulation became very low (about 10% on peaks) with normal gain settings on line amplifier. Checked with studio who reported 100% on peaks. Changed to spare line amplifier which resulted in normal modulation of transmitter. Checked all tubes on regular line amplifier (limiter). All OK except 6J7 in power supply very weak. Replaced with new tube and returned to regular operation. All OK. (We were operating on unusually low line voltage at the time. Not sure that this or defective 6J7 was entire trouble.)

Oct. 19-1947. Time 7:30 A.M. Sign-on delayed approximately 2 minutes due trouble in one of jacks on speech input bay. Everything checked OK about 2 minutes before sign-on time. Trouble apparently caused by pulling plug out of jack after making usual before sign-on test. Suggest jack contacts may be dirty or in need of adjustment.

Oct. 21-1947. Time 7:08 A.M. About 10 seconds after applying high voltage to transmitter, rectifier line overload occurred, along with plenty of arcing which appeared to be near bottom of modulator tube No. 2 and on north side. Search for defective parts revealed bad crack in insulator running from top to bottom. This insulator was the feed-through between filament transformer (farthest north) and the 872 rectifier farthest north. Replaced insulator. No further occurrence.

Dec. 13-1947. Time 6 to 6:30 A.M. Used auxiliary transmitter until 6:29::30 A.M. due to burned-out transformer in modulator bias power supply. Installed new transformer and put regular transmitter in at 6:29::30 A.M. Believe trouble was caused by a crack in the glass of one of the 866's, which tube was replaced. Believe this trouble could be eliminated in the future by all operators having modulator glass door unlocked until operator can observe what happens to above 866's when modulator door is locked.

Dec. 13-1947. Time 8:05 A.M. Frequency monitor started giving very erratic readings of approximately 5 low to 15 high. Replaced both 6H6 tubes, also 6F8, but did not help. New 6J5 cleared trouble at once.

Dec. 15-1947. Time 9:30 P.M. Frequency monitor was erratic again tonight. Cause uncertain.

Dec. 16-1947. Time 11 A.M. Replaced 5T4 in frequency monitor.

Jan. 7-1948. Diode adjust button on frequency monitor is in urgent need of attention. Sends pointer off-scale and upon return. readings are erratic.

Feb. 18-1948. Frequency monitor went bad at 4:55 P.M. Notified Chief Engineer at 7:30 P.M. Monitor removed from rack at 10 P.M. Found broken wire at meter range switch. Back in service at 11:05 P.M. OK.

March 19-1948. Time 11 P.M. Carrier off approximately 30 seconds due power line failure. Transmitter switched to stand-by line and program resumed.

April 5-1948. Time 9:21 P.M. Lightning burned coil out on relay used to short few turns in r-f coil for directional operation. Antenna currents unbalanced until operator located trouble and propped relay shut at 9:47 P.M. Entire relay replaced after sign-off and operation checked.

April 18-1948. Time 7:40 P.M. Off air approximately one minute.

A-c overload blew main line fuses. De-ion circuit breakers did not trip.

April 25-1948. Time 8:30 P.M. At 8:30::15 and at 8:35 P.M. A-c rectifier overloads blew main line fuses. Lost approximately 45 seconds on each occasion. After sign-off, found one well-done bug lying under one of the main rectifiers, which could account for one outage.

June 2-1948. Lightning took relay out in south tower just prior to going directional. Relay was blocked short manually, no time lost. Antenna meter, same unit, found to be reading low after sign-off, replaced both items. Relay now OK. Meter reads 0.3 low but meter was so tagged in box. Old meter burned up.

July 1-1948. Time 8:06::30 P.M. Program level dropped off suddenly at 8:06::30. Modulation about 5%. Notified Control where all OK. Changed to emergency line at 8:08::30, program level normal. Immediately checked regular line on spare line amplifier and it seemed clear. Returned to regular loop at 8:10 P.M. OK at this time. Notified local test board but could get no information. Only possible source of trouble I can think of other than line is the patch panel jacks, since regular limiter amplifier used with emergency line operated OK.

July 4-1948. Time 11:24 A.M. Regular transmitter off air 11:24 A.M. due failure of 892 final power amplifier. Auxiliary transmitter on air 11:25::30 A.M. Defective tube replaced and regular transmitter back on air at 11:40. The 892 removed had been increasingly harder to drive the last two or three hours.

Aug. 4-1948. Time 11:56 A.M. Carrier off 27 seconds. While replacing shorted 807 in speech amplifier section of transmitter. Program had been mushing up off and on all morning. Trouble traced to 807 stage after watching meter readings for quite a while. I noticed that when plate current would start to creep from 120 to over 130 mils, it would be followed by drop in level; mushing up and current would flicker according to noise. Replaced both 807's and found one to show a short. The other tested OK.

Sept. 29-1948. Time 6:00 A.M. Before sign-on, modulation monitor meter was kicking around and noise was definitely on carrier. Changed 807's and trouble cleared up. Checked tubes on checker and they checked OK. Cause of noise uncertain. These tubes were just put in service last month and should be OK.

Oct. 1-1948. After Sign-off. It was impossible to get the proper grid drive to final all evening. Tried replacing tubes in exciter. Ended up with a new 802 in the buffer stage, 805 in intermediate, and two 805's in exciter PA stage. Now all currents and drive are normal.

Oct. 13-1948. Time 6:49::30 A.M. Cut carrier for one minute to change 4-845 tubes and 2-807's. Program was mushing up and distorted at times. Before sign-on I noticed 845's plate current would kick up and return to normal. Don't know if this was caused by 845's or to something in preceding 807 stage. 807 currents were high again. Checking 807's on tube checker could find nothing wrong as was case with last pair removed. Suggest something might be wrong in screen voltage divider.

Oct. 28-1948. Currents on 807's in transmitter are now reading high again. Slight audio distortion noticeable on high levels.

Nov. 14-1948. After Sign-off. Replaced resistors *R20, R21, R22,* and *R23* in 807 screen-dividing network. One of these resistors seemed to be changing resistance under load. Suggest this as possible cause of previous trouble in 807 stage.

Nov. 18-1948. Time 6:30 P.M. Off air 6:30-6:36 P.M. to replace burning capacitor in south tower turning house. Noticed extremely high plate current on final (not enough to trip overload) and excessive common input meter reading at 5:55 P.M. Checked at tower house and found the burning capacitor. Reduced power to 1 kw. Called chief engineer 6:02 P.M. who arrived at 6:20 P.M. to assist in replacement. Back on air at 6:36 with 1 kw, back to 5 kw at 6:43 P.M. All OK. The large capacitor had broken down dielectric.

Nov. 18-1948. Time 8:37 P.M. Lost all line power at 8:37 P.M. Off air 20 seconds while changing power lines. Power on this emergency line very low. Notified service department at Power & Light Co. at 8:40 P.M. Returned to regular power line at 11 P.M. Right oscillator would not hold in oscillation properly apparently due to extremely low line voltage. Changed to left oscillator which held OK until 10:09 P.M. when it also went partially out providing less than half proper grid drive to final stage. Changed back to right oscillator which holds fairly well. They drop out intermittently but come back under actuation of the oscillator change-over switch. All OK after return to regular power line and normal voltage.

March 3-1949. Time 8:15-9:15 A.M. First noticed carrier breaks about 8:15. They were very short (about one-half second each) at first and far apart, increasing in frequency and length of duration. Changed 805 tubes in final of exciter at 8:48::30 to 8:49. This did not stop trouble. Went to auxiliary transmitter 8:52. Put in new 892 final amplifier and back on regular transmitter at 9:15 A.M. Trouble was caused by intermittent open grid connection inside of 892.

March 26-1949. Time 6:20 P.M. One of two lights on second level of north tower out at time of observation, 6:20 P.M. CAA was notified at 6:31 P.M. CAA requests notification on completion of repairs.

March 30-1949. Time 10:40 A.M. Replaced bad lamp socket on north tower light that was defective. Called CAA to report lamp back in.

May 17-1949. Time 6:14::30 P.M. Carrier off 10 seconds at 6:14::54 to 6:15::04 due dead tube #1P2 in final of exciter (2-805's). Tube #1P1, taking all the load, was running red hot. Thought it not advisable to hold off to 6:30 (sustaining program) as tube seemed to be intermittently shorting out (exciter final cathode current dropping almost to zero with tube running red, and frying noise). Four seconds (opening) of 6:15 program lost. Control cut chimes on preceding program saving several seconds.

June 29-1949. Time 6 A.M. Unable to get transmitter on before sign-on time because of bad circuit breaker in high-voltage rectifier a-c line. Had I been more observing I would have seen that pilot light for this switch was not on. I did not suspect this for the rectifiers were glowing weakly just as though there was very little or no load on them. All control and interlock circuits seemed complete and when main rectifier switch was thrown on, the final plate voltage and current meters would barely move upward and then return to zero. In working circuit-breaker switch up and down, suddenly, something snapped inside and power light came on. Did not have time to service it before sign-on time.

June 29-1949. After Sign-off. A-c rectifier line circuit breaker repaired. Trouble was two loose bolts. These bolts hold the lever mechanism. They were loose enough so as to allow the trigger to jump out of the slot and not trip the contact arm. Found the same bolts in the other circuit breakers one-half turn loose. If it ever becomes necessary to order one of these breakers, it should be an exact physical replica otherwise a major operation will be necessary to install it.

Aug. 16-1949. Time 4:08 P.M. Oscillator drawing high plate current and frequency wavering between +5 and +15. Changed to "Left" oscillator which was normal and stable. After sign-off, replaced 802 oscillator in "Right" stage. OK.

Sept. 10-1949. Time 8:00 P.M. Modulation monitor failed at 8 P.M. Percentage meter rested at 80% full scale. Trouble: two very weak 76 tubes and one shorted 1-V rectifier. Replaced above tubes.

Sept. 13-1949. Time 8:49 P.M. Off air 8:49-8:50 P.M. On auxiliary transmitter 8:50 P.M.-10 P.M. Back on regular at 10 P.M. Cause: Carrier went off with a puff of smoke from filter rack behind transmitter. In checking, found holding coil of filter capacitor shorting relay was open. Removed a-c contactor K-1 and K-2 on filter rack terminal board. Checked open but not shorted to ground. Strapped contacts K-3 and K-4 together to complete control circuits.

Oct. 28-1949. Time 7:52 A.M. Transmitter went off air due to breakdown of Faradon output coupling capacitor. Extinguished blaze around defective capacitor before putting auxiliary transmitter on. Time lost: 2½ minutes. Changed back to regular transmitter at 8:27 A.M. Time lost: ½ minute.

Nov. 29-1949. Time 5 P.M. Replaced one 89 in monitor amplifier. Tube was soft and monitor level low and distorted.

Jan. 11-1950. Time 9:31-9:56 A.M. Power-line failure caused us to lose one phase of three-phase supply on both regular and spare lines. Power failed at approximately 9:31 A.M. At first thought fuses were blown. Checked and found one phase very low. Switched power lines with same result. Tried to get auxiliary transmitter on air by blocking contactors but not enough voltage to operate passably. Reported to Power & Light Co. Service resumed at 9:56 A.M. Power & Light Co. says line crew got a short across line somewhere.

March 10-1950. Time 4:35 A.M. Applied carrier at 4:35 A.M. Noticed that plate current in power amplifier was too high, and that all r-f meters in phasing unit were too low. Also heard frying noise inside phasing unit like contacts arcing. Checked and found them OK. Noticed smoke coming from small fixed capacitor (0.0012) just below relay. Replaced it with 0.0015, as near as we had.

March 21-1950. Time 5:41::30 P.M. Bad distortion on program after 5:41::30 P.M. due sour 892 in final. One side of filament opened up. Grid drive very low and 805's in exciter were overloaded badly. Studio held off from 5:44::22 to 5:44::45 to change to auxiliary transmitter. On auxiliary 5:44::45 to 5:59::55. Changed back to regular at 6:00::05. Approximately 5 seconds of opening commercial lost. Bad 892 is #N1605. Replaced with #F0974.

April 23-1950. Time 3:47 P.M.-4:24::30 P.M. At 3:47 P.M. had direct lightning hit, burning out the antenna coupling capacitor 1-C-10, the common input meter, shorting the south tower line meter and also putting a short on the phasing unit for the north tower. Replaced the C-10 capacitor, and found that the phasing unit was so hopelessly shorted I bypassed it with a heavy copper strip, tying the output of transmitter directly to input of the south tower transmission line. Returned to air with low power at 4:20 P.M., high power at 4:24::30 P.M. Cleaned up shorts, replaced r-f meters and returned to normal operation through phasing unit at 6:40 P.M.

April 25-1950. Time 6:15 P.M. Going directional at 6:15 P.M., noticed practically no antenna current in north tower. Then arcing started in variable capacitor to right (from rear) in phasing cabinet. Reduced to low power nondirectional at 6:17-6:22 P.M. Found old piece of wire and solder in capacitor by turning all the way out of mesh. Returned to proper place (69 on dial). Back to normal high power directional at 6:22 P.M.

Chapter 16

WHY PREVENTIVE MAINTENANCE

THE PRIMARY PURPOSE of any preventive maintenance schedule, of course, is to reduce as much as possible the likelihood of failure during the broadcast day of any component part of the broadcasting installation. Regular maintenance schedules are in force at most broadcast stations, and do much to increase their useful life and anticipate and prevent many tube and parts failures that would occur if neglected.

Preventive maintenance of any sort of equipment may be defined as a systematic series of operations performed periodically on the equipment in order to prevent breakdowns. This type of maintenance may be divided into two phases: work performed while the equipment is functioning and work performed during the normal shutdown periods. Here we are concerned only with the shutdown period preventive maintenance.

The importance of preventive maintenance cannot be overestimated. The owners of a broadcast station depend upon its being on the air every second of its scheduled periods of transmission. It is of the utmost importance that the personnel of radio stations properly maintain their equipment so that lapses in the transmission will be kept to a minimum.

Cleanliness of equipment is of utmost importance since collection of dust and dirt has been known to cause a number of troubles. This is particularly true in the higher power stages of transmitters, since accumulation of foreign matter over a period of time reduces the voltage insulation to a point where leakage currents and arc-overs are common. High-voltage contacts have an extreme tendency to collect dirt (this is the principle used in electronic smoke eliminators), and the higher relative humidity existing in summertime or southern locations tends to aggravate this characteristic. A dusting and clean-up procedure, then, is a desirable nightly procedure at a transmitter plant. A source of dry air under pressure is a common means of blowing out dirt, dead insects, and the like from inaccessible corners and variable tuning capacitors. Insulators, safety gaps, etc. should be polished with a dry cloth or carbon tetrachloride used to loosen excessive dirt and grime.

Proposed Transmitter Maintenance Schedule

In order that preventive maintenance be effective, it is essential that it be performed at regular intervals; that is, certain portions of the equipment must be inspected for certain things every day while other parts of the equipment need only be inspected weekly or monthly in addition, of course, to those things which are inspected daily. Below will be found a comprehensive maintenance schedule[1] which may be considered as a guide to anyone desiring to set up such a means of preventing breakdowns. Naturally, items may be included in this schedule which may be felt to be unnecessary at some particular transmitter, but it has been compiled with the thought that every precaution should be taken.

Later on in this chapter will be found the actual preventive maintenance schedule which is followed at Station WIRE.

TRANSMITTER MAINTENANCE

A. DAILY

1. Hourly read all meters and check power tube filament voltages.
2. Check air-cooled anode temperatures. Check water temperature of water-cooled tubes.
3. Check for correct cabinet temperature of air around high-voltage rectifiers.
4. After shutdown make a general inspection for overheated components, such as capacitors, inductors, transformers, relays, and blowers.
5. Investigate any peculiarities of meter readings.
6. In the event of overloads, examine safety gaps and transmitter components for arc pits, etc. Clean and repolish surfaces where arcs have occurred. Reset gaps if necessary. Investigate cause of outages.
7. In the event of lightning or heavy static discharges, inspect the transmission line, terminating equipment, and antenna including gaps. Polish pitted surfaces.
8. If gas filled co-ax is used, check pressure.

[1] By courtesy of RCA Mfg. Co.

B. WEEKLY (In addition to above)

1. Immediately after shutdown, check antenna terminating components for signs of overheating.
2. Clean antenna tuning apparatus. Check for arc pits, etc. Clean and polish gaps and adjust if necessary.
3. Test antenna monitor rectifier tubes.
4. Calibrate remote antenna meters against meters in the antenna.
5. Clean transmitter with vacuum.
6. Clean component parts of transmitter.
   a. Brush terminal boards,
   b. Clean insulators with carbon tet,
   c. Clean power tubes and high voltage rectifiers with tissue and alcohol (or distilled water).
7. Check filament voltages and d-c voltages at the tube socket of all tubes which are not completely metered by panel meters.
8. Check air flow interlocks for proper operation. Check all door interlocks for proper operation.
9. Check operation of grounding switches. Examine mechanical operation and electrical contacts.
10. Inspect blowers for loose impellers, free rotation, and sufficient oil.
11. Inspect relays for proper mechanical and electrical operation. If necessary, clean and adjust components.
12. Inspect air filters; clean if excessive dirt has accumulated.
13. Check all sphere and needle gaps. Clean any pits or dirt. Check gap spacings.
14. Check filter bank surge resistors with ohmmeter.
15. Check any power tube series resistors with ohmmeter.
16. Check power change switches if used; check for no serious arcing during day-night antenna change-over if used.
17. Make general performance check-up. Distortion, noise, and frequency response. Observe modulated wave form on CRO.
18. Check neutralization by cutting crystal oscillator and observing grid currents. Observe overmodulation waveform envelope on CRO.
19. Check proper operating voltage for pure tungsten filament tubes. Operate at lowest voltage permissible as indicated by:
    a. AM transmitters—distortion and carrier shift checks.
    b. FM transmitters—decrease filament voltage until output begins to drop.

c. Operate filaments approximately 1% above filament voltage determined in a or b.

20. If water cooling is employed check entire system for any signs of leakage and for electrical leakage.
21. Check pressure of any gas-filled capacitors.

C. MONTHLY (In addition to above)

1. Make detailed inspection of all transmitter components with whatever tests of parts that may seem advisable.
2. Clean and inspect all vacuum and rectifier tube socket contacts, and the tube pins.
3. Clean air filter or replace. Brush dust from blower impellers, canvas boots, etc.
4. Clean and adjust all relay contacts. Clean pole faces on contactors. Replace badly worn contacts.
5. Oil blower motors (carefully).
6. Operate all spare vacuum tubes for a minimum of two hours under normal operating conditions. Clean up any gassy tubes.
7. Operate all spare mercury vapor rectifiers normally, after first applying filament only for a minimum of 30 minutes. Store tubes upright.
8. Inspect all variable inductor contacts for tension, signs of overheating, and dirt. Clean and adjust as required. Carbon tet or crocus cloth may be used for cleaning. Do not use emery cloth.
9. Check for proper operation of time delays, notching relays and any automatic control systems.
10. Clean audio equipment (console, etc.) attenuator and low level switching contacts with cleaner; wipe off excess.
11. Check tubes in station monitor equipment, such as frequency monitor, modulation monitor, etc.
12. Clean switches in monitoring equipment with cleaner.

D. QUARTERLY (In addition to above)

1. Lubricate tuning motors and inspect for ease of rotation.
2. Check all indicating meters (a-c, d-c, r-f). Check a-c filament voltmeters with an accurate dynameter type of meter.
3. Check *all* connections and terminals for tightness.
4. Inspect any flexible cables to door connections.
5. Inspect and lubricate if necessary any flexible drive cables.
6. Inspect, clean, and service (if necessary), all switches. Volt-

meter selector switches, push button switches, control switches, etc.

7. Clean transmission line insulators and take up slack if open wire lines are used.
8. Check oil circuit breakers, if used, for sufficient oil and loose or defective parts.

E. SEMIANNUALLY (In addition to above)

1. Test transformer oil for breakdown and filter it if necessary (power company).
2. Check protective overload relays or circuit breakers for correct operation.
    a. A-c overload relays may be checked by shorting the high-voltage transformer secondary.
    b. D-c overload relays may be checked by shorting the d.c. through the relay in the circuit protected by the relay.

MAINTENANCE SUGGESTIONS

A. CONTACTORS—GENERAL

1. Inspect all parts at regular intervals.
2. Parts should be kept free of dirt, grease, and gum.
3. Replace contact tips as needed (keep spares on hand).
4. Keep all contacts and interlocks clean and free from burrs and pits.
5. Main copper contacts should not be lubricated. Darkened tips due to overheating, or copper beads should be dressed with a fine file. (Do not use emery cloth.)

B. CONTACTORS—HUM

1. Clean off rust, dirt, or grease from pole piece and armature and apply a small quantity of light machine oil to prevent rusting.
2. Check pole shader and its circuit. Armature contact surfaces above and below shader should be approximately equal.
3. Armature to pole piece contact should be made over a large area; gaps should not be over 1/1000 to 2/1000 inch. If the contact area is small or the gap too wide, the pole face may be ground or filed down to a FLAT surface.

C. CONTACTS—SILVER

1. If not burned or pitted they may be cleaned with a contact burnishing tool.
2. If burned and pitted, dress with a small fine file and polish with crocus cloth.

Maintenance of Water-Cooling Systems

As has been stated before, it is not the purpose of this section to duplicate otherwise available data on the engineering aspects of transmitting equipment. However, since cooling systems of transmitters of more than 1-kw rating are so highly important to the problem of keeping the station on the air, some factors of operating and maintaining these cooling systems that have not been emphasized before will be presented here for the operator's convenience.

Fundamentally, the power rating of a tube in free air is determined by three characteristics, namely:

1. Plate voltage that may be safely applied (dependent on physical parameters).
2. Electron emission of filament.
3. Amount of heat that can be dissipated at anode without causing overheating (dependent on physical and electrical parameters).

Thus it becomes apparent that insofar as the operator is concerned, the third characteristic is the only variable in the problem of heat dissipation, since electrical parameters such as operating angle in electrical degrees, bias voltage, etc. are more or less under the direct supervision of the operator.

In water-cooled systems, the temperature of the water at the outlet of the tube jacket should never exceed 70° C. (158° F.) as indicated by the water thermometer at that point. The rate of flow should be approximately 15 gallons per minute; 20 gallons per minute is more beneficial in retarding accumulation of foreign matter in the jacket and the prevention of steam bubbles along the anode surface. The effect of increasing the flow of water is to increase the turbulency of flow. This increased turbulency breaks down the layer of steam present at the anode wall and increases the heat exchange between the wall and the water. The turbulency of the flow may be increased by mechanical means, such as baffles.

Extraordinary precautions must be taken in the installation of the tube in the water jacket. The movable metal parts of the jacket should be coated with a light film of oil to help prevent corrosion. The tube should then be placed gently in the jacket, and after it is correctly seated the retaining studs or jacket clamping device is fastened firmly into place to force the flange of the plate into solid contact with the watertight gasket. The electrical connections may then be made. Care should be taken that the wires are not near or do not touch the glass bulb. Should this precaution be neglected, puncture of the glass from corona discharge is likely to occur. Particular care should also be observed in making the connection between hose and jacket tight and clean. Because of electrolysis, trouble is likely to develop at this point, and close inspection every two or three weeks is advisable.

A reasonably rigid maintenance schedule should be observed on the entire system to forestall trouble from water leakage, scale formation, or the formation of steam bubbles with resultant transmitter shutdown and loss of time on the air. Leaks, of course, may be temporarily sealed to a certain extent by using friction tape until permanent repairs can be made after sign-off. In some instances, it is possible to cover the radiator used to cool the water with a blanket until the inlet temperature of the water rises to around 104° F., resulting in a slight expansion of the parts which will aid in sealing a minor leak.

Scale formation, if and when it occurs, will prevent adequate transfer of heat from anode to water. If it becomes necessary to remove the tube for this or any other reason, the tube should be lifted carefully from the jacket after the clamping device has been released. Sticking of the tube often occurs, and in this case a gentle twisting back and forth while lifting will free the tube. Immersion of the plate in a 10% solution of hydrochloric acid is usually recommended to dissolve scale formation. The anode should then be rinsed thoroughly in distilled water.

The formation of steam bubbles may be checked periodically by using a good insulating rod at least six feet long. This should be moved along the jacket while aural observations are made. Precautions should be taken to assure the operator's safety, including grounding the testing tube between water jacket and the observer by a "hot stick" or similar arrangement.

A convenient way for the operator to keep an approximate check on the heat dissipation of the tube is by use of the formula:

$$P(KW) = \frac{n\,(t_o - t_i)}{4}$$

where:

t_i = known initial temperature of water in degrees C
t_o = temperature of water at jacket outlet in degrees C
n = rate of flow in gallons per minute.

It should be remembered here that the filament heat is also being dissipated into the water. It is recommended that the operator read the manufacturer's instructions that come packed with transmitting tubes. Some of the foregoing information is from RCA tube instruction sheets.

Forced-Air Systems

Although the problems of deteriorating hose, leaky hose connections, electrolysis, and troublesome flow-interlocks have been largely overcome by porcelain reels, reliable flow-interlocks and completely nonferrous circulating systems (all-copper tank and pipes), the familiar problems have remained of scale formation, gradual water evaporation, and relatively large time consumption in changing tubes.

In recent years, the elimination of the water-cooling system has been accomplished for transmitters up to and including 50-kw rating by the development of forced-air cooling systems. Control circuits for this system are greatly simplified, consisting as they do of an air-interlock damper on top of the blower motor, which prevents application of filament and plate voltages until normal air-flow pressure is present, and a blower motor "keep-alive" relay, which is a time-delay relay keeping blower motors functioning 4 to 7 minutes after filament voltage is removed.

Maintenance of forced-air systems is simpler than that of water systems but is just as important for trouble-free operation.

The canvas air ducts should be cleaned about once a month by removing them, turning them inside out, and using a vacuum cleaner to remove accumulated dirt. While these ducts are removed, a cloth may be used to slide between the fins of the tube, especially in against the tube anode, to remove dust. Care should be taken not to damage the mercury air-flow switches which are mounted on the blower housing. These switches prevent the application of filament and plate voltages until proper air flow is present. Both sides of the air-flow vanes (half-circle disks used to operate the mercury switch) should be wiped clean with a cloth or chamois and a small wire brush may be used to clean the corners of the fan blades. A vacuum cleaner then should be used to pick up any dust from inside the bottom of the blower frames.

After this cleaning procedure has been carried out, the blowers should be started to check the air-flow vanes for proper operation of the mercury switches, canvas ducts replaced, and over-all operation checked.

STATION WIRE PREVENTIVE MAINTENANCE SCHEDULE

The following preventive maintenance schedule has been in use at Station WIRE and as may be seen from designations, it is planned to have this work done on each Thursday night after the transmitter goes off the air. This arrangement is different from the general schedule presented earlier in this chapter inasmuch as some of the maintenance operations are performed weekly, others monthly, and the last group is performed five or six times per year.

First Thursday Night of Month Maintenance Schedule

Before turning main transmitter off, after conclusion of program from studio, read both North and South Tower Antenna Current meters. Check and adjust Remote-Reading Antenna meters, on the operating console, to read with their respective Tower meters. Turn main transmitter off in accordance with the usual sign-off procedure.

Shift antenna change-over switch to the Auxiliary Transmitter side. Turn on Auxiliary Transmitter for one-half hour check. Make regular transmitter log on this half-hour operation. Also, enter the readings of both the Antenna Current and Remote-Reading Antenna meters on this log sheet.

In addition to the daily sign-off procedure, carefully remove the two oscillator tubes and wipe them free of dust.

Clean both sides of plate glass partitions in front of the three large tubes. Use damp cloth on this and dry with paper hand towel.

Open all glass meter partition doors at top of transmitter and clean glass on both sides. Clean all glass meter faces including those on the power panel and Audio racks.

After Auxiliary Transmitter has been tested, turned off and log completed, BE SURE TO RETURN ANTENNA CHANGE-OVER SWITCH TO THE 5-D OPERATING SIDE.

Vacuum—clean the top of transmitter cabinets.

Wipe off all insulators on top of transmitter and insulators holding copper bus bar to rear wall, and all insulators on top of phasing cabinet (to transmission line), and all insulators on modulation transformer and reactor.

Wipe off all insulators and all components on the filter rack (filter condensers, reactors, resistors, relays, etc.).

Remove all rectifier tubes (remember to keep them upright in case of mercury-vapor rectifiers) from their sockets. Remove the shields and clean the sockets and other components under shields.

Open all front interlock doors. Use vacuum to clean all reachable space from the front, behind center panels, tube shelves, and floor plates.

Wipe off the four high-voltage insulators between modulator blower motors and the insulator above each of the high-voltage rectifiers. Also the insulators above the first and second audio stage tubes. Wipe off insulators on condenser above power-amplifier blower motor. Also the rectifier and oscillator feed-thru insulators in bottom of Exciter Unit.

Clean all components in modulator section ABOVE the final-tube holders. Inspect and tighten all connections in this half section.

Open rear doors on modulation and final stage. Wipe off floor of both from the rear. Wipe off all three blower frames. BE VERY CAREFUL OF ALL INSULATORS, ESPECIALLY MICALEX.

Take and record reading of "Filament Elapsed Hours Clock."

REPORTS: Make Transmitter Operating Room Report, as shown in Fig. 16-1, that this schedule was completed or any deviations from it. Also any other observations made.

TRANSMITTER OPERATING ROOM REPORT

DATE 8-15-1950 TIME MAINTENANCE PM

PROGRAM AFTER SIGN OFF MAINTENANCE

ANNOUNCER

CONTROL OPERATOR ORIGINATED

TRANSMITTER OPERATOR

REMARKS:

MAINTENANCE SCHEDULE COMPLETED IN ENTIRETY.

NOTE: ONE 807 TUBE (OUR NUMBER ")_ 206) REMOVED. WEAK.

BY H. Innes

(FILE IN DUPLICATE WITH CHIEF ENGINEER)

Fig. 16-1. An example of a transmitter operating-room report as used by the author at Station WIRE.

FINAL CHECK: Turn on transmitter in usual manner, first on low power of 1000 watts. If everything is normal then check the 5000-watt operation.

Second Thursday Night of Month Maintenance Schedule

Remove canvas air ducts beneath modulator and final tubes and place paper or cloth cover over blower openings. Carefully clean between fins of all three large tubes by sliding cloth between fins especially in against tube anodes.

Clean all components in lower half of modulator section. Check all components in this section including terminal blocks.

Remove temporary cover over blower openings. Wipe off both sides of air flow vanes. Check for free movement.

Brush corners of fan blades with small brush, then vacuum.

Some dust usually remains in bottom of blowers and should be removed by running each blower and using a deflector over blower opening on top to direct air away from tube bases. After all three are cleaned, and with blowers running, check to see that the air-flow vanes are operating mercury switches properly.

Replace canvas air ducts, and double-check for proper cond.

Run regular Auxiliary Transmitter test.

REPORTS: Make Transmitter Room Report that this schedule was completed or any deviations from it. Also report if conditions of air filters on back doors necessitates replacement.

FINAL CHECK: Check for low power and high power operation in usual manner.

Third Thursday Night Maintenance Schedule

Regular Auxiliary Transmitter check.

Clean relay contacts in phasing unit.

Inspect and clean lightning gaps on transmission line above phasing unit.

Tighten and clean all connection and chassis of tuning assemblies in tower houses. Clean relay contacts.

Check relay operation for Directional and Non-Directional operation by turning transmitter on as outlined for previous maintenance schedules.

Fourth Thursday Night Maintenance Schedule

Regular Auxiliary Transmitter check.

Clean input and output attenuators on 96-A line amplifier, also monitor attenuator. (Use Lubriplate and clean cloth.)

Vacuum jack strips on speech input panel.

Inspect for tightness connections on relays in power-control panel. Clean the above relay contacts.

Wipe off filament rheostats on power control panel.

Wipe off all accessible places on power control panel.

Clean "modulator bias" relay contacts.

Clean the contacts of the two overload relays and time-delay relay in exciter unit.

Clean and inspect all components in exciter unit not covered in previous schedules.

Vacuum inside of power-control console, tighten all connections.

Regular transmitter check for normal operation.

Fifth Thursday Night Maintenance Schedule (where this occurs)

Regular Auxiliary Transmitter check.

Check all tubes (with tube checker) of 96-A line amplifier and associated power supply, modulation monitor, frequency monitor, program monitor, and speech input tubes.

Check 6K7 balance in 96-A line amplifier.

Check spare line amplifier for proper operation.

Check lubrication of all motors including toilet and shower water pumps and exhaust fans.

Take inventory of new transmitter log sheets, transmitter-room report sheets, spare tubes, spare fuses, indicator lamps, and illuminating lamps for building and towers.

Chapter 17

PREVENTIVE MAINTENANCE INSTRUCTIONS

IN THE PRECEDING chapter some general facts about preventive maintenance were presented together with schedules showing when the different operations should be performed. Inasmuch as some of these operations deal with apparatus that can easily be damaged unless proper care is exercised, certain procedures should be followed so that no damage does result from the periodical inspections and so that if repairs to the apparatus are necessary, they can be effected properly. It should be borne in mind that the data in the following pages are general for the most part and it may be that some manufacturers recommend specific procedures for their products, which, of course, should be followed.

The reasons why preventive maintenance operations are followed seem obvious with no further comment. It might be well, however, for the men who are responsible for this maintenance work to keep in mind that the procedures discussed in the following pages have been designed to

1. Combat the detrimental effects of dirt, dust, moisture, water, and the ravages of weather on the equipment.
2. Keep the equipment in condition to insure uninterrupted operation for the longest period of time possible.
3. Maintain the equipment so that it always operates at the maximum possible efficiency.
4. Prolong the useful life of the equipment.

Preventive Maintenance Operations

The actual work performed during the application of the preventive maintenance schedule items is divided into six types of operations. Throughout this section, the lettering system for the six operations is as follows:

F—Feel	C—Clean
I—Inspect	A—Adjust
T—Tighten	L—Lubricate

a. Feel (F). The "Feel" operation is used most extensively to check rotating machinery (such as blower motors, drive motors, and generators) for over-heated bearings. "Feeling" indicates the need for lubrication or the existence of some other type of defect requiring correction. Normal operating temperature is that which will permit the bare hand in contact with the motor-bearing cover for a period of 5 seconds without feeling any discomfort. The "Feel" operation also is applied to a few items other than rotating machinery; the "Feel" operation for these items is explained in the discussion of each specific item.

Note: It is important that the feel operation be performed as soon as possible after the shutdown, and always before any other maintenance is done.

b. Inspect (I). "Inspection" is probably the most important of all the preventive maintenance operations. If more than one man is available to do this work, choose the most observant, for careful observation is required to detect defects in the functioning of moving parts and any other abnormal conditions. To carry out the "Inspection" operation most effectively, make every effort to become thoroughly familiar with normal operating conditions and to learn to recognize and identify abnormal conditions readily.

"Inspection" consists of carefully observing all parts in the equipment. Notice such characteristics as their color, placement, and state of cleanliness. Inspect for the following conditions:

(a) Overheating, as indicated by discoloration, blistering or bulging of the part or surface of the container; leakage of insulating compounds; and oxidation of metal contact surfaces.

(b) Placement, by observing that all leads and cabling are in their original positions.

(c) Cleanliness, by carefully examining all recesses in the units for accumulation of dust, especially between connecting terminals. Parts, connections, and joints should be free of dust, corrosion, and other foreign matter. In tropical and high-humidity locations, look for fungus growth and mildew.

(d) Tightness, by testing any connection or mounting which appears to be loose, by slightly pulling on the wire or feeling the lug or terminal screw.

c. Tighten (T). Any movement of the equipment caused by transportation or by vibration from moving machinery may result in loose connections which are likely to impair the operation of the set. The importance of firm mountings and connections cannot be overemphasized; however, *never* tighten screws, bolts, and nuts unless it is definitely known that they are loose. Fittings that are tightened beyond the pressure for which they were designed will be damaged or broken. When tightening, always be certain to use the correct tool in the proper size.

d. Clean (C). When the schedule calls for a "Cleaning" operation, it does not mean that every item which bears that identifying letter must be cleaned each time it is inspected. Clean parts only when inspection shows that it is necessary. The "Cleaning" operation to be performed on each part is described later on.

e. Adjust (A). Adjustments are made only when necessary to restore normal operating conditions. Specific types of adjustment are described later.

f. Lubricate (L). Lubrication means the addition of oil or grease to form a film between two surfaces that slide against each other, in order to prevent mechanical wear from friction. Generally, lubrication is performed only on motors and bearings.

Note: When a part is suspected of impending failure, even after protective maintenance operations have been performed, immediately notify the person in charge who will see that the condition is corrected by repair or replacement before a breakdown occurs.

Suggested List of Tools Necessary for Relay and Commutator Maintenance

A number of items on the preventive maintenance schedule require work of a special and somewhat delicate nature. This work includes cleaning and repairing relay contacts, cleaning plugs and receptacles, polishing commutators, and adjusting motor and generator brushes. To do the work properly, special supplies and specially constructed tools are needed. A suggested list is given below:

Nonmagnifying dental mirror.
Cleaning brush, 2-inch.
Canvas-cloth strip.
Sandpaper strip, fine.
Sandpaper strip, semifine.

Crocus-cloth strip.
Small relay crocus-cloth stick.
Relay-contact burnishing tool.
Fine-cut file.
Brush seating stone.
Commutator polishing stone.
Canvas-cloth stick.
Crocus-cloth stick.
Sandpaper stick.
1 Brush, cleaning, 1-inch.
1 Brush, cleaning, 2-inch (2).
1 Carbon tetrachloride, quart can.
2 Cement, household, tube.
1 Cloth, canvas, 2 x 4-feet.
1 Cloth or canvas, strip, 2 x 6-inch, cut from sheet (3).
1 Cloth, lint-free, package.
6 Crocus-cloth, sheets.
1 Crocus-cloth, strip, ¾ x 6-inch, cut from sheet (6).
1 File, small, fine cut.
1 Lubricant, Vaseline, container.
1 Mirror, nonmagnifying dental.
6 Sandpaper sheets, #0000.
6 Sandpaper sheets, #00.
1 Sandpaper, #0000, ¾ x 6-inch, cut from sheet.
1 Sandpaper strip, #00, ¾ x 6-inch, cut from sheet.
1 Stick, crocus-cloth, large.
1 Stick, special, canvas-covered.
1 Stick, special, crocus-cloth stick for relays, small.
1 Stick, special, sandpaper covered.
1 Stone, commutator polishing stone.
1 Stone, brush seating stone.
50 Tags, small marker.
1 Tool, relay contact burnishing.

Construction for Relay and Commutator Tools

Crocus-cloth, canvas-cloth, and sandpaper sticks are constructed in the following manner:

1. First prepare a length of wood 3¾ inches long, ⅜ inch wide, and 1/16 inch thick or less. Cut one piece of crocus cloth 2½ inches long and 1 inch wide.

2. Fold the crocus cloth as in Fig. 17-1 (A) and cement it to the stick. Note that both sides of the stick are covered. Place the stick in the vise, press it and wait until the cement hardens. Cut off the piece of crocus cloth which extends over the edge of the stick.

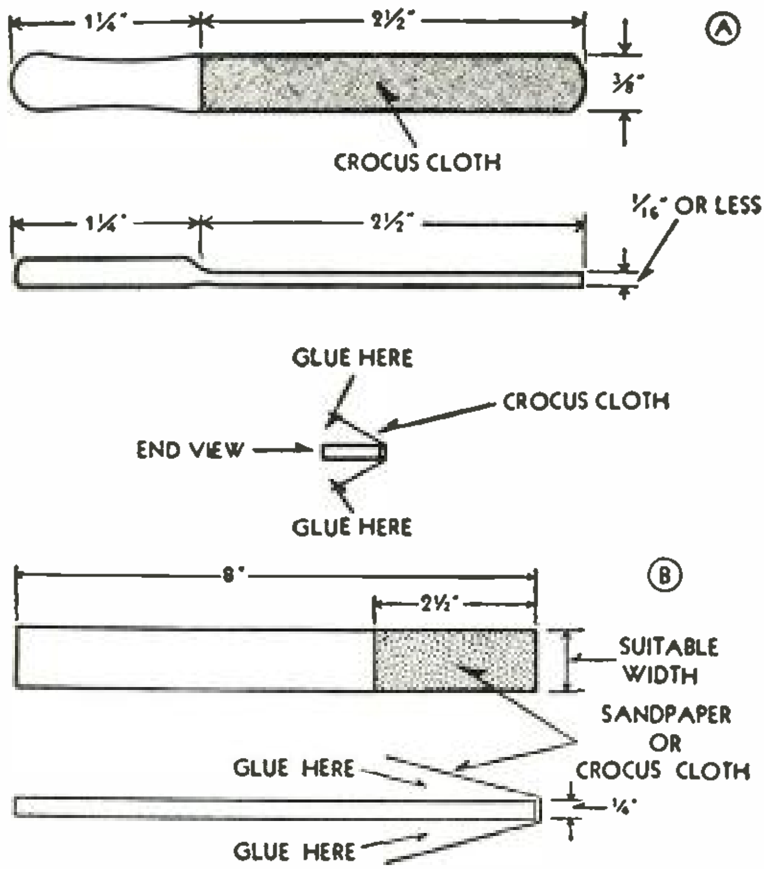


Fig. 17-1. The crocus-cloth stick (A) is used for cleaning relay contacts and the one in (B) is for cleaning motor or generator commutators.

3. Obtain three pieces of wood which measure 8 inches long, 1 inch wide, and approximately ¼ inch thick. Cut one piece of crocus cloth, one piece of #0000 sandpaper, and one piece of canvas cloth, each 5¼ inches long and 1 inch wide.

4. Fold the long, narrow pieces of crocus cloth, sandpaper, and canvas cloth as shown in Fig. 17-1 (B) and cement one of them to each of the three sticks. Note that in this case the fold is over one end of the stick rather than over the sides. Place the sticks in the vise, press, and wait until the cement hardens.

Use and Care of Tools

Proper care of tools is as necessary as proper care of radio equipment. Any effort or time spent in caring for tools is worth while. Clean them when necessary and always replace them so that they are easily accessible. The following information will be helpful in using and caring for the tools listed below.

Crocus-Cloth Stick. The crocus-cloth sticks are used to clean contacts of relays in the radio equipment.

Large Commutator Sticks. Commutator sticks with covering of sandpaper or canvas are used for cleaning commutators of electric motors and generator sets.

Commutator Dressing Stone. The dressing stone is used only in case of emergency to dress a commutator or motor generator.

Brush Seating Stone. The seating stone is used when a set of new brushes is installed in alternators or exciters. Only a very limited application of the seating stone is required to seat the average set of brushes.

Electric Soldering Iron. The use of the soldering iron is generally known. Remember to keep the tip properly tinned and shaped.

Allen Wrenches. Allen wrenches are used to tighten or remove the Allen setscrews on fan pulleys, motor pulleys, etc. These are small wrenches and should be kept in the cloth bag provided for that purpose. After use, wipe them off with an oily rag, replace them in the bag, and restore them to the tool box.

Diagonal-Cutting Pliers. Diagonal pliers are used to cut copper wire (no larger than No. 14) when working in small places. Do not cut iron wire with the diagonals.

Gas Pliers. Gas pliers are used to hold round tubing, round studs, or any other round metal objects that do not have screw driver slots or flat sides for wrenches.

Long-Nose Pliers. Long-nose pliers are used to hold and dent small wires and to grip very small parts. They are generally used around delicate apparatus.

Adjustable End-Wrenches. Adjustable end-wrenches are designed to remove or hold bolts, studs, and nuts of various sizes. Keep the adjusting-nut free from dirt and sand and oil them frequently.

Nut-Driver Wrenches. Nut-driver wrenches are used to remove nuts of various sizes. Choose a wrench that fits the nut snugly.

Screw Drivers. Screw drivers of different sizes are important tools and must be kept in good condition. Select the proper size for the job to be done. Never force a screw; if undue resistance is felt, examine the threads for damage and replace the screw if necessary.

Shorting Bar. The shorting bar must be constructed at the station. Obtain a piece of wood about 15 inches long and 1 inch thick. Fasten a piece of copper or brass rod or tubing securely to one end of the stick in such a manner that the rod extends 12 inches beyond the end of the stick. Solder a piece of heavy flexible wire about 18 inches long to the metal rod at the point where it is fastened to the stick and attach a heavy clip to the free end of the wire. When using the shorting bar, always attach the clip to a good ground connection BEFORE making contact with the terminal to be grounded.

Vacuum Tubes

The purpose of tube maintenance is to prevent tube failures caused by loose or dirty connections and to maintain the tubes in a clean operating condition at all times. Certain types of vacuum tubes, especially those used in high-voltage circuits, operate at high temperatures. Careless contact with the bare hands or arms causes severe burns. Sufficient time must be allowed for the tubes to cool before handling.

Maintenance of vacuum tubes involves making minor adjustments and cleaning. Tubes requiring the most frequent maintenance are those used in high-voltage circuits. Because of their high operating potentials, these tubes require more frequent inspection and cleaning than tubes used in low-voltage circuits. Loose coupling at the terminals of high-voltage tubes will result in the terminals becoming pitted and corroded. Loose connections cause poor electrical contact and lower the operational efficiency of the unit in which they are employed.

Maintenance of vacuum tubes should be applied only when necessary. Too frequent handling may result in damage to the tube terminals and connections. As a rule, vacuum tubes need little maintenance; therefore, when the program calls for maintenance, but inspection shows that the tubes do not require it, the operation should be omitted. It is advisable, however, to clean the glass envelopes of the tubes and remove dust or dirt accumulations surrounding their immediate areas. The *object* of the maintenance program is to maintain the tubes free from dirt, oil deposits, and corrosion.

Vacuum tubes for maintenance purposes are divided into two groups:

(1) Transmitting-type tubes.

(2) Receiving-type tubes.

Maintenance procedures required for vacuum tubes differ according to types. Certain maintenance operations that must be performed on transmitting-type tubes may be omitted in the maintenance of receiving-type tubes. Transmitting-type tubes are those used in transmitters, modulators, and high-voltage rectifier units. Because of their physical construction they require careful inspection and cleaning during maintenance.

Five procedures are required to the performance of maintenance of vacuum tubes: feel, inspect, tighten, clean, and adjust. The procedures involved depend on the type of tube being maintained. Transmitting tubes may require the application of the above-mentioned procedures, while the procedures required for receiving tubes are limited by tube types.

Maintenance Procedures

The following procedures should be employed for the maintenance of vacuum tubes:

Caution: Discharge all high-voltage capacitors before performing any maintenance operations. Avoid burns by allowing sufficient time for tubes to cool before handling.

Feel (F). (1) This operation should be applied only to high-voltage tubes, such as those used in transmitters, modulators, and high-voltage rectifier units.

Note: The following operations should be performed 5 to 10 minutes after power has been removed from the tubes.

(2) Feel the grid, plate, and filament terminals of the tubes for excessive heat. Practice will determine the temperature to be accepted as normal. For example, when two grid terminals are felt, one should not be warmer than the other. Excessive heat at terminals indicates poor connections.

Inspect (I). This maintenance operation is applicable to all types of vacuum tubes and should be performed after the tubes have had sufficient time to cool.

(1) Inspect the glass or metal envelopes of tubes for accumulations of dust, dirt, and grease. Inspect the tube caps and connector clips for dirt and corrosion. Inspect the complete tube assembly and socket for dirt and corrosion. Check the tube caps to determine whether any are loose. On glass tubes, check the glass envelope to determine whether or not it has become loosened from the tube base. Replace tubes which have loose grid caps or envelopes when these

faults are discovered. If replacement is impossible, do not attempt to clean or handle the tube, operate the tube as it is, providing that its operation is normal. Enter the tube condition in the log so that replacement can be made at the earliest possible time.

(2) Examine the spring clips that connect to the grid plate, and filament caps for looseness. Also examine all leads connected to these clips for poorly soldered or loose connections. These leads should be free of frayed insulation and broken strands. When removing clips from loosened grid caps, extreme care must be exercised, particularly if corrosion exists. Never try to force or pry a grid clip from the grid cap of a tube as damage to the tube or grid cap may result. If the grid cap is loose and it is necessary to remove the grid clip, first loosen the tension of the clip by spreading it open; then gently remove (do not force) the clip from the tube cap.

(3) Inspect the tubes to be sure they are secure in their sockets. Certain types of receiving tubes used are mechanically fastened with tube spring locks; others have sockets in which the tube itself is locked in place. Inspect by turning the tube in clockwise direction in its socket until it is locked in place. This type of socket is generally used for the transmitting-type tubes. However, the firmness with which the tube is held in place depends upon the tension of the terminals in the socket. These terminals are of the spring type (contact springs) and must have sufficient tension to make good contact against the tube prongs. The tension can be tested by grasping the tube and turning it first counterclockwise and then clockwise to its original position. If the tube seems to snap into place as it is turned, the spring tension of the socket terminals is firm enough; however, if the tension seems weak, they may be tightened or adjusted as explained in the tube maintenance procedure under "Adjust."

(4) Inspect all metal tubes for signs of corrosion and looseness of mounting. Many receiving-type tubes have keyways in the center of the tube bases. These keyways sometimes become broken, and have a tendency to loosen the tube in the socket. Do not attempt to replace tubes that have broken keyways unless it is absolutely necessary to do so, and it is possible to replace the tube correctly in its proper position. Inspect the tube sockets of metal tubes for cracks or breaks. Do not force metal tubes into their sockets. If they are hard to replace, examine the tube pins for signs of corrosion or solder deposits.

Tighten (T). (1) In performing this operation, take care not to overtighten tube sockets, tube clamps, and tube socket insulators.

Porcelain sockets and stand-off insulators crack due to heat expansion if they are excessively tightened. Do not overtighten them. Care should be taken when tightening the tube caps of high-voltage tubes. Use the proper screw driver or tool; if the tool should slip it may fall against the glass envelope of the tube and ruin a perfect tube.

(2) Tighten all tube connections, terminals, sockets, and stand-off insulators which were found loose during the inspection procedure. When tightening tube sockets having stand-off insulators, determine before tightening whether the fiber washers between the socket and the stand-off insulators are intact. If these fiber spacers are cracked or missing, replace them before tightening the tube socket. Tightening the socket without these spacer washers breaks or cracks the porcelain-tube socket.

Clean (C). In the performance of this item, clean only where necessary. Do not remove tubes for cleaning purposes unless it is impossible to clean them in their original positions. If the tube must be removed, exercise care in doing so. Do not attempt to clean the envelopes if they are located in an out-of-the-way place; in this case remove them for cleaning. When tubes are removed for cleaning, replace them immediately afterward. Do not leave them where they may be broken.

(1) Clean the entire tube assemblies with a clean dry cloth if the glass envelope is excessively dirty. Wipe the glass envelope with a damp cloth moistened in water. Polish after cleaning with a clean dry cloth. Do not wipe metal tubes with a cloth moistened in water, as this causes the metal body of the tube to rust. Use a cleaning agent if the tube is excessively dirty because of oil deposits. Generally, metal tubes with oil deposits on their envelopes can be cleaned successfully by polishing dry with a clean dry cloth. The oil film remaining on the metal body of the tube prevents the tube from rusting. To remove oiliness, corrosion, or rust from tube envelopes, moisten a clean cloth with cleaning agent and clean the area affected until it is clean. Wipe dry with a clean dry cloth.

(2) Clean the grid and plate caps, if necessary, with a piece of #0000 sandpaper, or crocus cloth. The paper should be wrapped around the cap and gently run along the surface. Excessive pressure is not needed; neither is it necessary to grip the cap tightly. Clean the caps completely before replacing them on the tube terminals if corrosion is noted on the grid or plate caps.

(3) When the tube sockets are cleaned and the contacts are acces-

sible, fine sandpaper should be used if corrosion is present on the contacts. Clean the contacts thoroughly after sandpapering. Clean all areas surrounding tube sockets with a brush and a clean dry cloth; this prevents dust and dirt from being blown back on the tube envelopes when the unit is put back into operation.

Adjust (A). When performing this operation, care must be taken to arrange all leads and terminals to correspond as closely as possible with their original positions.

(1) Adjust all leads and tube connections. Check to determine if the leads are resting on the glass envelope of the high-voltage tubes; if they are, redress the leads so that proper spacing is obtained. Examine all leads connecting to the tube caps. These should not be so tight that they barely reach the caps of the tubes. If this condition is found, redress these leads so that enough "play" is obtained. Adjust all grid clamps so that the proper tension is obtained. To increase the tension of tube clamps, close the spring clamps slightly with a pair of long-nose pliers until the proper tension is obtained. Do not flatten the clamps.

(2) Tube sockets used for transmitting-type tubes should be adjusted if the tube is found loose in its mounting. The terminals of these sockets are spring-tensioned so that they may be adjusted to increase the pressure against the tube pins. To adjust these contacts, simply bend them toward the center on the socket until the correct tension is obtained. Do not apply too much pressure to the spring contacts; they may be broken from their mountings in the porcelain socket.

(3) Any difficulty in removing or inserting metal tubes can be remedied easily. Remove the metal tube and examine the tube pins to determine if solder or corrosion has accumulated on the pins. Remove solder deposits with a penknife; then polish the pins with fine sandpaper. Do not use a soldering iron to remove solder deposits; this makes them worse, as the solder is built up on the pins rather than removed. To remove corrosion, use fine sandpaper, but never use it unless it is absolutely necessary. Saturate a small piece of cleaning cloth with light lubricating oil or petroleum jelly, and wipe the tube pins. Remove the excess oil from the pins by wiping them almost dry with a clean dry cloth. If these procedures are followed, no difficulty will be experienced in removing or reinserting the metal tube into its socket.

Caution: Do not force metal tubes into their sockets. Do not pry or "wiggle" them loose, since this damages the prongs of the socket and results in intermittent operation of the unit in which they are located.

Special Instructions for Transmitting-type Tubes

High-power transmitter tubes and mercury-vapor rectifiers require special consideration in addition to the preceding as follows.

Mercury-Vapor Rectifier. As soon as possible after receipt of a new batch of mercury-vapor rectifiers, they should be placed in the tube sockets of the transmitter *without* connecting the anode lead. Filament voltage should be applied and left on for at least 30 minutes to distribute the mercury properly. They should then be placed in a rugged container *upright*, and prevented from jarring or tipping which would splatter the mercury. Should this occur, they must again be seasoned before application of plate voltage. Applying anode voltage to an unseasoned mercury-vapor rectifier will cause severe arc-backs within the tube.

While on the subject of arc-backs, the maintenance technician should become familiar with the most reliable means of checking the condition of a mercury-vapor rectifier. This may be done most reliably by the use of a cathode-ray oscilloscope.

Obtain a 115-volt 1 to 1 isolation transformer with a rating of at least 300 volt-amperes. Leave the tube in its regular socket, with anode disconnected but filament voltage applied. With one side of the isolation transformer connected to the 115-volt line, connect the secondary between the anode of the rectifier and the filament center-tap. To load the tube properly, a 200-watt, 50-ohm resistor should be connected in series with the voltage source. The oscilloscope is connected from anode to filament center-tap, making the connection directly to the cathode-ray tube without amplifiers. The sweep should be self-synchronized with the source.

The idea is to inspect the firing characteristic of the mercury-vapor rectifier with the 115-volt test voltage. A faulty tube fires later and later in the conduction alternation (half-cycle) and displays a greater reverse-voltage transient peak.

Fig. 17-2 illustrates examples of patterns obtained. The trace is a reproduction of the full voltage wave across the tube, the nonconducting alternation being sinusoidal, the conducting alternation trace showing the instantaneous voltage drop during conduction. This trace also shows the peak arc-drop, the sharp wavefront transient at the start of the conduction half-cycle which is an important characteristic in rectifier performance.

An excessive arc-drop is what causes the backfiring under operation. When this transient is little more than the normal 10-volt drop of the tube as in (A) of Fig. 17-2, the tube is good. When it reaches close to 40 volts (a calibrated screen should be used on the oscilloscope), the tube is doubtful. A trace showing a 60-volt arc-drop is bad and should definitely be discarded.

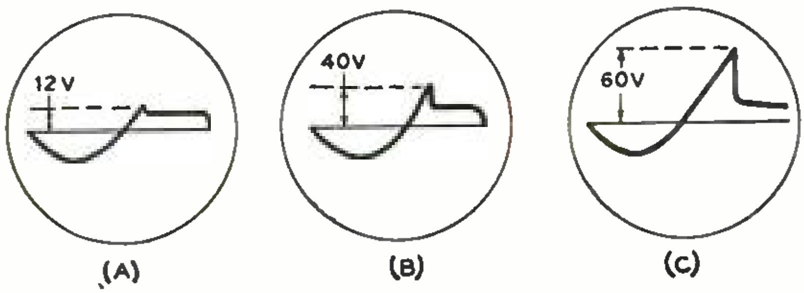


Fig. 17-2. Oscilloscope patterns obtained for good and bad mercury-vapor rectifier tubes. The arc drop shown in (A) is normal, the drops shown in (B) and (C) are excessive.

Large Power Tubes. The larger-type power tubes such as used in modulator and final stages in a transmitter of 5 kw or more, also require special treatment upon receipt from the factory, and at 4 to 6 months intervals thereafter. They should be placed in the transmitter and filament voltage (only) applied for 30 minutes. Low plate voltage should then be applied for about 15 minutes. This processing aids materially in preventing gas formation within the tube, which is a common occurrence if such maintenance measures are not followed.

Occasionally a tube will develop a small amount of gas while in storage. RCA recommendations for their RCA 892-R tube are as follows. With the tubes in the power amplifier, apply low plate voltage without modulation. After a few minutes apply 1,000 cps tone modulation, gradually increasing the percentage modulation. If no gas flashes occur after 15 minutes of full tone operation, remove the modulating signal, apply high power, then repeat above process.

If gas flashes occur during the process, go back to the low-power position with no modulation and repeat. Allow the tube to run for a considerable length of time with a low percentage of modulation, and then repeat the foregoing procedure.

An RCA 892-R tube that develops a small amount of gas in modulator service may usually be cleaned up by operating it in the power amplifier.

Capacitors

High-Voltage Capacitors. High-voltage capacitors, because of their high operating potentials, must be kept clean at all times to prevent losses and arcing. Dirt, oil deposits, or any other foreign matter must not be allowed to accumulate on the high-voltage terminals of these capacitors. All leads and terminal connections must be inspected periodically for signs of looseness and corrosion, and the porcelain insulators inspected for cracks or breaks.

Low-Voltage Capacitors, Oil-Filled. Low-voltage oil-filled capacitors require the same care as those of the high-voltage type, although the frequency of the maintenance operation is not so critical. The terminals and connections of these capacitors should be given the same careful inspection as those of the high-voltage types. The leads of these capacitors are not as rugged as those used on the high-voltage capacitors and should be inspected more closely for poorly soldered connections.

Tubular Capacitors. These capacitors are of the low-voltage paper type and are generally used in low-voltage circuits for coupling and bypassing. They should be inspected and cleaned whenever the chassis in which they are located is removed for maintenance. The only maintenance requirement for these capacitors is inspection of the tubular body of the capacitor for bulging, excessive swelling of the capacitors, and for signs of wax leakage. The terminal leads (pigtail type) of the capacitors are inspected for firmness of contact at their respective points of connection. Never use a cloth to clean this type of capacitor, as damage to the surrounding circuits may result. These capacitors are easily cleaned with a small, soft brush. All dirt and dust are brushed from the body of the capacitor and the surrounding area.

Mica Capacitors. Mica capacitors require very little maintenance other than being kept free from dust and oil. Two types of mica capacitors are generally used: the high-voltage and the low-voltage type. The low-voltage types are inspected whenever the chassis of the unit in which they are located is being maintained. The capacitors are inspected for cracked body conditions caused by excessive heat, while their leads (pigtail type) are inspected for firmness of contact at their respective points of connection. The high-voltage types, however, require terminals because of their high operating potentials. These terminals must be inspected for tightness and corrosion, firmness of mounting, and body conditions. The body of these capacitors is of a ceramic material and care must be exercised when tightening the mountings of these capacitors. The bodies of the capacitors are easily kept clean with a dry clean cloth. For satisfactory operation the terminals must be free from dirt and corrosion at all times. Take care when tightening the terminals of these capacitors, as excessive pressure damages or cracks the ceramic case where the terminals are coupled to the body of the capacitors.

Trimmer Capacitors. In very damp climates, trimmer capacitors must be inspected often. Dampness, if allowed to accumulate on the plates of the capacitors, results in erratic operation of the unit in which the capacitors are used. In certain cases where high voltage is used, serious damage to the capacitors results. A minute amount of moisture or a tiny bead of water is all that is necessary to short-circuit the plates of the capacitor and cause abnormal operation. When such conditions are encountered, the capacitor must be thoroughly dried by the heat process which requires the use of a small portable heater. A cleaning cloth used to dry the plates of the capacitors may throw the plates out of alignment when the cloth is inserted between them. In extreme cases where the plates of the capacitors are very closely spaced, use a magnifying glass to locate the exact position of the moisture beads existing between the plates. Due to the sheen of the capacitor plates, very minute particles of moisture cannot always be detected by the naked eye.

Maintenance Procedures for Capacitors.

Caution: To avoid severe electrical shock in case of bleeder failures, *discharge* all high-voltage capacitors before maintenance.

Feel (F). Feel the terminals of the high-voltage filter capacitors. These should be fairly cool. Excessive heat probably indicates losses due to loose, dirty, or corroded terminal connections. Feel the sides of oil-filled and electrolytic capacitors. These should be cool or slightly warm. If they are very warm or hot, the condition indicates excessive internal leakage. Capacitors in this condition are subject to failure at any time and should be reported for immediate replacement.

Inspect (I). Inspect the general condition of all capacitors regardless of type. Inspect for broken, frayed, or loose terminals, leads, and connections. Inspect the condition of the terminals of the high-voltage capacitors. Check these for dirt, corrosion, and looseness. Inspect the body of the capacitors for excessive signs of bulging and oil leakage.

Inspect the plates of the tuning capacitors for dirt and corrosion. Check all capacitor shafts, bushings, bearings, and couplings for looseness or binding.

Tighten (T). Tighten all loose terminals, connections, and terminal leads on all types of capacitors. Tighten all capacitor mountings and stand-off insulators. Tighten all loose shaft couplings and bushings.

Clean (C). Special attention should be given to all high-voltage capacitors to insure that they are not only kept clean, but are free from moisture. Thoroughly clean the insulators, terminals, and leads of high-voltage capacitors. When extremely damp, due to high humidity, these capacitors frequently have to be wiped dry with a clean, absorbent cloth to prevent arc-overs and breakdown of insulation. Remove terminals that appear to be either corroded or dirty; also remove those causing power losses due to high-resistance connections. Clean them with a crocus cloth which is either dry or moistened with cleaning fluid. Polish the terminals dry after cleaning with a clean, dry cloth. Replace all connections after cleaning, making certain that good electrical contact is obtained. The low-voltage capacitors require little attention. However, all insulated bushings and supports should be kept clean and free from foreign matter.

Adjust (A). Adjust all leads if necessary. This requires the redressing of leads which may have been dislocated during the maintenance procedure. If capacitor leads are stretched too tightly, redress or replace them until the correct lead placement is obtained.

Resistors

Resistors may be divided for maintenance purposes into two groups: the first group consists of those resistors easily detachable and known as *ferrule-type resistors;* the second group includes those whose terminals are soldered and are known as *pigtail-type resistors.*

a. Ferrule-Type Resistors.

Caution: Do not touch power resistors immediately after the power has been shut off. They are usually hot, and severe burns may result.

Feel (F). The springiness of ferrule clips may be ascertained when removing the ferrule-type resistor. Insufficient *pull* at the clip may be an indication of a loose connection and poor electrical contact.

Inspect (I). It is important to inspect all types of resistors for blistering or discoloration, for these are indications of overheating. Inspect the leads, clips, and metalized ends of the resistors and adjacent connections for corrosion, dirt, dust, looseness, and broken strands in the connecting wires; also inspect the firmness of mounting.

Tighten (T). Tighten all resistor mountings and connections found loose. If the tension at the end clips has decreased, it is common practice to press the clip ends together by hand or with a pair of pliers. The hand method is preferred because the pliers may bend the clip or damage the contact surface.

Clean (C). Dirty or corroded connections of ferrule-type resistors can be cleaned by using a brush or cloth dipped in cleaning fluid. If the condition persists, use crocus cloth moistened with cleaning fluid. It may be necessary to sandpaper the resistors lightly with fine grade sandpaper, such as #0000. Always wipe clean with a dry cloth before replacing them. Vitreous resistors connected across high voltage should be kept clean at all times to prevent leakage or flashovers between terminals. They should be wiped clean with a dry cloth or a cloth moistened with cleaning fluid. If cleaning fluid is used, the resistors must be polished with a dry clean cloth.

Pigtail-Type Resistors. Maintenance of pigtail-type resistors is limited to an inspection of soldered connections. Such connections may break if the soldering is faulty or if the resistors are located in a place subject to vibration. The recommended practice is to slide a small insulated stick lightly over the connections and to inspect them visually for solidity. If connections are noticeably weak or loose, they should be re-soldered immediately. Discolored or chipped resistors indicate possible overloads. Although replacement is recommended, resistors in this condition have been known to last indefinitely. The pigtail-type connections should be dusted with a brush or with an air blower if available.

Fuses

A fuse consists of a strip of fusible metal inserted in an electric circuit. When the current increases beyond a safe value, the metal melts, thus interrupting the current. Fuses vary in size and rating depending upon the circuits at which they are used. Some are designed to carry currents in milliamperes. Being very rapid in action, they protect the equipment from overloads and damage. Two types of fuses are used: renewable and nonrenewable. The first type is designed so that the fuse link, or element contained within the fuse cartridge, may be removed and replaced when blown. The second type, however, is constructed so that the fuse element is permanently sealed within the fuse housing. When a fuse blows, an attempt must be made to determine the reason for its failure, and to make corrections, if possible, before a new fuse is installed; then the complete fuse assembly must be replaced.

Renewable Type. The renewable type fuse assembly consists of a housing or cartridge of insulating material with a threaded metal cap (ferrule) at each end. The fuse element or link, as a precaution against damage, is placed inside the cartridge or housing and it is held in position by the two end caps, or ferrules. When a fuse is placed in service, the two ends of the fuse cartridge are slid into spring contacts mounted on the fuse block. This places the fuse in the circuit to be protected.

Nonrenewable Type. When nonrenewable fuses are blown, they must be discarded. Certain types of nonrenewable fuses are removed by unscrewing and withdrawing the cap screws that hold them in place. When removed, the fuse and cap screw are separated by pulling apart. The glass fuses are easily removed for inspection. Care must be taken to see that the fuse end and holding clips are kept clean and tight . If they are not, overheating will result and make replacement necessary.

Inspect (I). Inspect the fuse caps for evidence of overheating and corrosion. Inspect the fuse clips for dirt, loose connections, and proper tension.

Tighten (T). Tighten the end caps, the fuse clips, and connections to the clips on replaceable fuses if they are found to be loose. The tension of the fuse clips may be increased by pressing the sides closer together. Fuse caps should be hand-tightened only. Excessive tightening results in difficulty in removing them when required.

Clean (C). Clean all fuse ends and fuse clips with fine sandpaper when needed; wipe with a clean cloth after cleaning. If it becomes necessary to use a file to remove deep pits in the clips, fuse ends, or contacts, always finish up with fine sandpaper in order to leave a smooth contact surface. As a final step, wipe the surface clean with a clean dry cloth.

Bushings and Insulators

Bushings and insulators are extremely important elements in electric circuits, especially when located in high-voltage circuits where insulation breakdown is most common. Most of the high-voltage insulators are constructed of ceramic material with highly glazed surfaces.

Caution: Exercise extreme care when working near these insulators. They are easily chipped or broken.

Inspect (I). Thoroughly inspect all high-voltage insulators and bushings for moisture, dust, and other accumulated foreign matter. Unless they are both clean and dry, leakage or arc-overs will occur and damage them permanently. Check the chipped surfaces, hair line cracks, carbonized arc-over paths, and other surface defects that may make the insulator unserviceable. Insulators in this condition should be reported to the person in charge for replacement.

Tighten (T). Feed-through bushings, stand-off and other insulators should be tightened if found to have loose mountings or supports. Tighten these insulators with care because gaskets absorb only a small amount of pressure before breaking.

Clean (C). Cleaning operations are similar to those outlined for tubes. Use a clean cloth (dampened with cleaning fluid if necessary) to remove dust, dirt, or other foreign matter. Always polish with a dry, absorbent cloth after cleaning.

Relays

The various types of relays may be classified as follows: overload relays, time delay relays, and magnetic contactors. Relays require a certain amount of preventive maintenance, which must never be performed except when absolutely necessary. Certain types will be found to be completely encased in dustproof and moistureproof cases. These require little maintenance other than a periodic inspection.

Maintenance of relays requires that they be inspected periodically and preventive maintenance measures performed if necessary. The inspection procedure requires that the terminals be inspected for looseness, dirt, and corrosion. Contacts may have become loosened because of the jarring of the equipment during shipment. The contacts may become dirty or corroded due to climatic conditions where the equipment is being operated. Relay contacts must never be sandpapered or filed unless the operation is absolutely necessary for the normal operation of the relay unit. A relay is considered normal if:

(1) The relay assembly is free from dirt, dust, and other foreign matter.
(2) The contacts are not burned, pitted, or corroded.
(3) The contacts are properly lined up and correctly spaced.
(4) The contact springs are in good condition.

(5) The moving parts travel freely and function in a satisfactory manner. The solenoids of plunger type relays must be free from obstructions.
(6) The connections to the relay are tight.
(7) The wire insulation is not frayed or torn.
(8) The relay assembly is securely mounted.
(9) The coil shows no sign of overheating.

A relay is considered abnormal if it fails to meet any of the above-mentioned requirements. The following are the maintenance procedures used in the maintenance of relay units.

Inspect (I). Inspect the relays, to determine abnormal conditions using the check list given above. If the contacts are not readily accessible, they may be examined with the aid of a flashlight and mirror. Many of the relays can be inspected and cleaned without being removed from their mountings or without being taken apart. Mechanical action of the relays should be checked to make certain that the moving and stationary contacts come together in a positive manner and that they are directly in line with each other. The armature or plunger mechanism should move freely without binding or dragging. Care should be taken during inspection not to damage or misalign the relay mechanism. Relays that require the removal of the cover for complete inspection may be found enclosed in glass, Bakelite, or metal cases. Relays must never be taken apart unless it is absolutely necessary. If they must be taken apart for maintenance purposes, care should be exercised in doing so. When disassembling relays, tag all leads as they are being removed. This insures that the proper leads are returned to their proper terminals after the maintenance procedure is completed.

Tighten (T). Tighten all loose connections and mounting screws found loose, but do not apply enough force to damage the screw or to break the part which it holds. Do not start screws with their threads crossed. If a screw does not turn easily, remove it and start again. Relay coils can be tightened by inserting, if possible, a small wooden or paper wedge between the coil and the core of the relay. This prevents chatter of the relay. Tighten any and all loose connections. Tighten also the mounting of the relay assembly, if it is found loose. When replacing glass or Bakelite covers over relay cases, take care not to overtighten the screw cap holding the glass or Bakelite cover over the relay assembly.

Clean (C). Clean the exterior of the relay with a dry cloth, if it is very dirty; clean with a cloth or brush dipped in cleaning fluid; then wipe the surface with a dry cloth. If loose connections are found, they should be inspected. If inspection reveals that the connections are either dirty or corroded, they should be removed and cleaned before tightening.

The relay service aid is a narrow piece of folded cloth or canvas. It serves a twofold purpose: it is suitable for polishing a clean surface, and it is used as a follow-up to a crocus cloth. It is also intended to remove grains of pumice which came off the crocus cloth and adhere to the contact surface. The cloth is used as shown in Fig. 17-3.

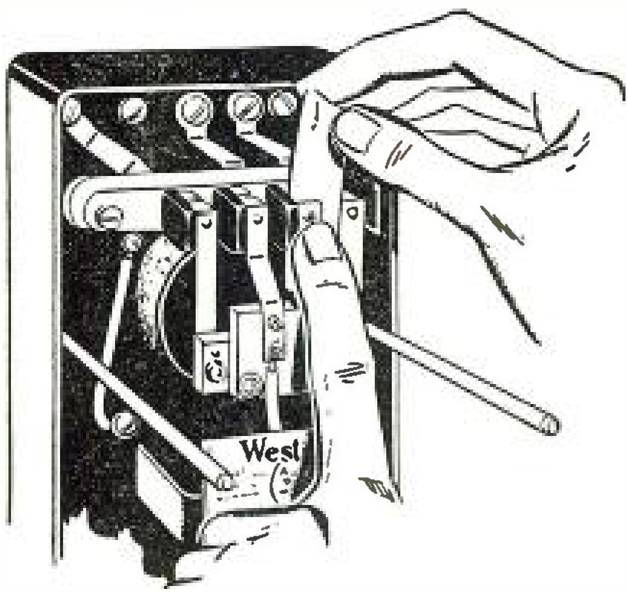


Fig. 17-3. Relay contact surfaces are polished by a narrow strip of cloth or canvas as shown in the sketch.

Cleaning Relay Contacts. The following information should be carefully studied. It instructs how relay contacts of various types should be cleaned.

Hard contacts. Hard alloy contacts are cleaned by drawing a strip of clean wrapping paper between them while holding them together. It may be necessary in some cases to moisten the paper with cleaning fluid. Corroded, burned, or pitted contacts must be cleaned with the crocus cloth strip or the burnishing tool as shown in Fig. 17-4.

Solid silver contacts. Dirty contacts. Dirty solid silver contacts are easily cleaned with a brush dipped in cleaning fluid. After being cleaned, the contacts are polished with a clean dry cloth.

Note: The brown discoloration that is found on silver and silver-plated relay contacts is silver oxide and is a good conductor. It should be left alone unless the contacts must be cleaned for some other reason. It may be removed at any time with a cloth moistened in cleaning fluid.

Corroded contacts: Dress the contacts first with crocus cloth, using either the stick or the strip of crocus material. When all of the corrosion has been removed, wipe with a clean cloth moistened in cleaning fluid and polish with a piece of folded cloth. Make certain that the shape of the contacts has not been altered from the original.

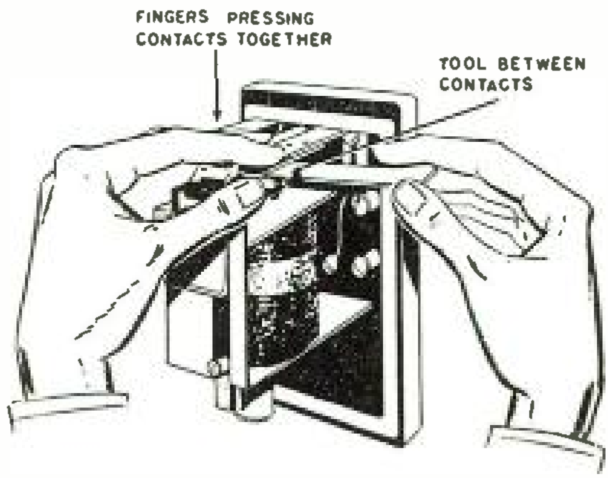


Fig. 17-4. Hard alloy contacts of a relay are cleaned by pulling a strip of clean wrapping paper between them while pressing the contacts together.

Burned or pitted contacts: Resurface the contacts, if necessary, with #0000 sandpaper, making certain that the original shape of the contacts is not changed. Next, smooth the surface of the contacts with crocus cloth until a high polish has been obtained. Wipe thoroughly with a clean cloth to remove the abrasive remaining on the contacts. When contacts are very badly burned or pitted and replacement is not available, the small fine-cut file and #0000 sandpaper should be used in keeping with instructions given later.

Silver-plated contacts. Dirty contacts: Dirty silver-plated contacts are cleaned with a cloth or brush dipped in cleaning fluid. After being cleaned, the contacts are polished with a dry cloth.

Corroded contacts: Dress first with crocus cloth, using either the stick or strip of crocus material. The work must be done very carefully not to remove an excessive amount of silver plating. When all of the corrosion has been removed, polish with a clean dry cloth. Make certain that the shape of the contacts has not been altered.

Burned or pitted contacts: Dress the contacts with crocus cloth until the burned or pitted spots are removed. This may require an appreciable amount of time and energy, but it is preferable to using a file or sandpaper. If it is found that the crocus cloth does not remove the burns or the pits, use the sandpaper tool very carefully. When sandpaper is used, it must be followed with crocus cloth to polish the contacts, and then wiped thoroughly with a cloth moistened in cleaning fluid. The contacts are then polished with a clean dry cloth.

Warning: Never use highly abrasive materials, such as emery cloth, coarse sandpaper, or carborundum paper for servicing relay contacts, as damage to the contacts will result.

Adjust (A). Adjust relay contacts after cleaning if necessary. The contacts should close properly when the plunger is hand operated. Adjust the relay springs if necessary. Do not tamper with the relay springs unless it is absolutely necessary. These springs are factory adjusted and maintain a certain given tension and rarely get out of adjustment. If the spring tension must be changed, exercise care when doing so. The adjustment of the current control relays is accomplished by setting calibrated knobs to the desired setting, or by turning a knurled adjustment sleeve which has a calibrated scale mounted adjacent to it. The adjustments should not be changed from their original factory setting except in cases of emergency. Overload relays must never be adjusted unless the person in charge has been notified, and has sanctioned the adjustment.

Shapes of Relay Contacts. Relay contacts are of varied shapes, as shown in Fig. 17-5 depending upon their size and application. In some

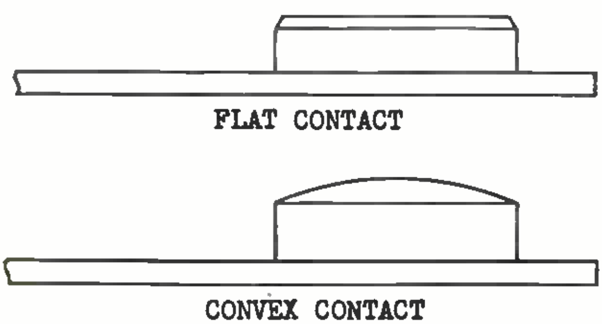


Fig. 17-5. The original shape of the contacts must be retained. This shape may be either flat or convex, as shown at the left.

instances, both contacts are flat; in others, one contact is convex while its mate is flat. The original shape of a contact must be retained during cleaning. If burning or pitting has distorted the contact so that it must be reshaped, the original shape must be restored. It is essential that the maintenance personnel familiarize themselves with all details of the relays by examining them while the relays are in good condition. In this way, they will be better prepared to do their work well.

Relay Servicing Tools and Their Use

To service the relay contacts, several types of tools are needed. Each of these has a special function, as described below.

The Burnishing Tool. This tool is used on relays which have extremely hard contacts made of palladium or elkonium. This tool is not a file. A contact should not be burnished unless it is found to be pitted or oxidized, and then not more than is necessary to restore a clean smooth surface. The original shape of the contact must be retained.

Small Fine-Cut File. This file is to be used only on the larger contacts when they have become very badly burned or pitted, and a replacement is not available. This tool is not to be used on silver-plated contacts, or on the contacts of the telephone-type relays. The file should not be used more than is necessary to remove the pit. The original shape of the contact must be preserved. After filing, #0000 sandpaper should be applied to the contact, and followed with crocus cloth to obtain a smooth finish on the contact surface. A clean dry cloth serves for final polishing.

The #0000 Sandpaper Stick. This tool is made in the same way as the crocus-cloth stick, except that sandpaper is used instead of crocus cloth. The use of sandpaper is limited, as is the use of the fine-cut file, to the treatment of badly burned or pitted contacts on the larger relays. Sandpaper is not used on silver-plated contacts, except under extreme circumstances, and when used should be followed with crocus cloth. All contacts should be polished after sanding, with a clean dry cloth.

Crocus Cloth. This maintenance aid is available in two forms—as a tool and as a strip of material. It serves a twofold purpose: it may be used to remove corrosion from all relay contacts, or it may be applied to the contacts following the use of the fine-cut file and #0000 sandpaper. Neither the file nor sandpaper leaves a finish smooth enough for proper relay operations. Use crocus cloth to polish the surface of the contact. The choice between the stick and the piece of cloth depends upon accessibility. If the location of the relay and the position of the contacts permit the use of the crocus-cloth stick, it should be used; otherwise, the strip of crocus cloth must serve. The crocus cloth and tool are used as illustrated in Figs. 17-3 and 17-4. In both cases the maintenance aid is inserted between the contacts and is drawn through them while the contacts are pressed together with the fingers.

Switches

For the purpose of maintenance, switches may be classified into two general groups: those whose contacts are readily accessible, and those whose contacts are completely encased. The basic maintenance operations of "Inspection," "Cleaning," "Adjusting," and "Lubrication" are applicable only to the first group. Because of the enclosed construction of the second group, no maintenance can be applied. The work is limited to a mechanical test of their operations.

Accessible Contact Switches. This group consists of knifeblade switches, start-stop push-button switches, and high-voltage shorting bars. With the exception of the shorting bars, all of these switches consist of blades which are mechanically inserted into spring contacts.

Inspect (I). Inspect all the terminal connections of each individual switch for tightness and cleanliness. The mounting of the switch should be checked for firmness. Operate the mechanism of the switch and see if the parts move freely. Observe the stationary spring contacts to determine whether they have lost tension and whether they are making good electrical contact.

Tighten (T). All loose mountings and connections should be tightened properly. If inspection shows that the fixed contacts have lost tension, tighten them with the fingers or pliers. Tighten every connection or terminal found loose.

Clean (C). If inspection shows that any terminal, connection, or section of the switch is dry, dusty, corroded, or pitted, clean the part by using a dry clean cloth. If the condition is more serious, moisten the cloth with cleaning fluid and rub vigorously. Surfaces which have been touched with the bare hands must be thoroughly cleaned with a cloth moistened in cleaning fluid, and then polished with a clean cloth. The points of contact with the moving blade are naturally those which most often show signs of wear. Examine these points very carefully to insure that both sides of each blade, as well as the contact surfaces of the clips, are spotlessly clean at all times. Crocus cloth moistened with cleaning fluid should correct this condition; however, if it is not corrected, #0000 or #000 sandpaper may be used. Always polish clean after the sandpapering operation.

Adjust (A). Some of the switches have a tendency to fall out of alignment because of loosening of the pivot. In most cases, tightening the screw on the axis of motion corrects this condition.

Lubricate (L). If binding is noted during inspection of the operation of the switch, apply a drop of instrument oil with a toothpick to the point of motion or rotation. Do not allow oil to run into the electrical contacts, as a film of oil may cause serious damage or a poor contact. Lubrication of switches is not recommended unless serious binding is noticed.

Nonaccessible Contact Switches. Under this heading are included all the remaining switches not discussed in the previous paragraph. Interlock switches, toggle switches, meter protective push buttons, and selector switches have been designed so that it is impossible to get at the contact without breaking the switch assemblies. The only maintenance possible is to check the operation of the switch assemblies and, if something abnormal is detected, to notify the person in charge immediately so that a spare may be obtained and a replacement made as soon as possible. Do not lubricate any of these switches under any circumstances.

Generators and Motors

Certain preventive maintenance procedures must be applied to these components if proper functioning and dependable performance are to be obtained. There are three principal cases that contribute to faulty operation of this type of equipment: accumulation of dirt, dust, or other foreign matter on the windings and moving parts of the equipment; lack of sufficient lubrication on bearings and other moving parts; and improper adjustments or damaged parts. Given proper maintenance care, motors and generators give long and efficient service. In addition to the techniques given in the following paragraphs, additional maintenance instructions covering certain motors or generators will be found in various items of the manufacturer's instruction books. Unless specifically mentioned, the maintenance techniques that follow apply to the motors and generators used in the transmitter.

Feel (F). The bearing and the housings should be tested by feeling them to determine overheated conditions. An accepted test, except in very hot climates, is to hold the bare hand in contact with the bearing or housing for a period of at least 5 seconds. If the temperature can be tolerated this length of time, the bearing temperature may be considered normal. Overheating may indicate lack of sufficient lubrication, a damaged bearing surface, or, in rare situations, an excessive accumulation of dirt within the field windings.

Inspect (I). Each motor and generator exterior, and any other visible parts, must be inspected for dirt and signs of mechanical looseness or defects. Wherever wires are exposed, see that all connections are tight and in good condition and that the insulation is not frayed. Inspect the motor ends for excess oil and the mounting for loose bolts. Wherever possible and practicable, feel the pulleys, belts, and mechanical couplings to insure that the proper tension or tightness is present.

Tighten (T). Any mounting, connection, or part found loose must be properly tightened. If any internal part such as a commutator segment or an armature coil appears loose, notify the person in charge and repair the part immediately or replace it at the first opportunity. Operation under these conditions will cause considerable damage in a very short period of time.

Clean (C). Carefully wipe the exterior, base, and mountings of each motor and generator with an oiled cloth in order to leave a thin, protective film of oil on the surfaces. If available, use an air blower, or hand bellows to blow the dust and dirt out when inspection shows that the windings are dusty or dirty.

If inspection of the commutator and brushes shows that cleaning is necessary, the accepted cleaning practice is as follows: lift or remove the most accessible brush assembly and press a piece of canvas cloth folded to the exact width of the commutator against the commutator; then run the motor for about 1 minute, exerting the necessary pressure. If the condition still persists because the commutator has been burned or pitted, use a piece of fine sandpaper (#0000), preferably mounted on the commutator cleaning stick, and, exerting the necessary pressure, rotate the motor for approximately 1 minute. Stop the motor and wipe around the commutator bars with a clean cloth. It may be necessary to polish the commutator with a piece of canvas, as explained in the first procedure. Identical maintenance procedures apply to slip rings.

Transformers and Choke Coils

Some transformers are enclosed in metal housings, others are external, but in all cases they are impregnated with insulating compound. As a result, similar maintenance techniques are applicable to all of them.

Inspect (I). Carefully inspect each transformer and choke for general cleanliness, for tightness in connections of mounting brackets and rivets, for solid terminal connections, and for secure connecting lugs. The presence of dust, dirt, and moisture between terminals of the high-voltage transformers and chokes may cause flashovers. In general, overheating in wax- or tar-impregnated transformers or coils, is indicated by the presence of insulating compound on the outside or around the base of each transformer or coil. If this condition is encountered, immediately notify the person in charge.

Tighten (T). Properly tighten mounting lugs, terminals, and rivets found loose.

Clean (C). All metal-encased transformers can be cleaned easily by wiping the outer casings with a cloth moistened with cleaning fluid. Clean the casing and the immediate area surrounding the transformer base. Clean any connections that are dirty or corroded. This operation is especially important on high-voltage transformers and coils. It is very important that transformer terminals and bushings on all types of transformers be examined and kept clean at all times.

Variacs

Variacs, as a rule, are built sturdily and are protected so that very little maintenance other than regular inspection is required.

Inspect (I). Carefully inspect the exteriors of the variacs for signs of dirt and rust. Inspect the mounting of each variac to determine whether it is securely mounted. Inspect all connections for looseness, corrosion, and dirt. Check the slip rings for signs of corrosion or dirt.

Clean (C). The perforated casing of each variac as well as the area surrounding the base must be cleaned regularly. If the slip rings need cleaning, dismount the variac and clean with a cloth moistened in cleaning fluid and then polish with a clean dry cloth. If the dirty condition persists, use crocus cloth and rub vigorously. Again polish with a clean cloth. Reassemble the variac; then reinstall it, reconnecting all terminals carefully.

Lubricate (L). If the variac shaft shows signs of binding or if it squeaks, apply a few drops of household oil to the front and rear bearings. Rotate the control shaft back and forth several times to insure equal distribution of the lubricant in the front and rear bearings.

Rheostats and Potentiometers

Rheostats and potentiometers fall into two main groups for maintenance purposes; those which have the resistance winding and the sliding contact open and accessible, and those which, by construction, have their inner parts totally enclosed. In the latter group, very little maintenance can be performed, since opening and removing the metal case may damage the unit.

Inspect (I). The mechanical condition of each rheostat must be inspected regularly. The control knob should be tight on the shaft. Inspect the contact arm and resistor winding for cleanliness and good electrical contact. Check the rheostat assembly and mounting screws for firmness; the sliding arm for proper spring tension; and the insulating body of the rheostat for cracks, chipped places, and dirt.

Tighten (T). Tighten carefully any part of the rheostat or potentiometer assembly found loose.

Clean (C). The rheostat or potentiometer assembly is easily cleaned by using a soft brush and then polishing with a soft clean cloth. If additional cleaning is needed, or if the windings show signs of corrosion or grease, the brush may be dipped in cleaning fluid and brushed over the winding and contacts. With a clean cloth, remove the film that remains after the cleaning fluid has evaporated. If the contact point of the sliding arm is found burned or pitted, it is good practice to place a piece of folded crocus cloth between the contact and the winding and then to slide the arm a number of times over the crocus cloth. When cleaning the winding, do not exert excess pressure, or damage will result.

Adjust (A). If the tension of the sliding contact is insufficient, an adjustment can be made with the long-nose pliers. Slight bending of the rotating piece in the proper direction restores the original tension.

Lubricate (L). Apply lubrication only when necessary; that is, when binding or squeaking is noticed. One or two drops of instrument oil applied to the bearings with a toothpick is sufficient. Since the slightest flow of oil into the winding or the sliding-arm contact may cause serious damage, lubrication should be applied very carefully and only on the bearings. Wipe off all excess oil.

Terminal Boards and Connecting Panels

Little preventive maintenance is required on terminal boards and connecting panels.

Inspect (I). Carefully inspect terminal boards for cracks, breaks, dirt, loose connections, and loose mountings. Examine each connection for mechanical defects, dirt, corrosion, or breakage.

Tighten (T). All clean terminals, screws, lugs, and mounting bolts found loose should be tightened properly. Use the proper rods for the tightening procedure and do not overtighten or the assembly may become cracked or broken.

Clean (C). If a connection is corroded or rusty, it is necessary to disconnect it completely. Clean each part individually and thoroughly with cloth or crocus cloth moistened with cleaning fluid. All contact surfaces should be immaculate for good electrical contact. Replace and tighten the connection after it has been thoroughly cleaned.

Air Filters

Air filters are placed in blowers and ventilating ducts to remove dust from the air drawn into and circulated through the ventilating system. Some filters are impregnated with oil and some are filled with cut strands of glass to facilitate the filtering action. The following procedures cover their maintenance:

Inspect (I). The filter should be inspected for any large accumulation of dirt and for lack of oil. Note whether the filter is mounted correctly and whether the retaining clips are in place. Improperly assembled filter elements or wall frames, allow unfiltered air to leak around the edges and thus permit dust to enter the ventilating system.

Tighten (T). Tighten the retaining clips if they are found loose, and readjust the filter in its mounting.

Clean (C). The filters are easily accessible and may be taken out after removal of the cover plate. The general procedure is, as follows: mark the outside of the filter before removing it from the air duct. Before washing it, tap its edges against the wall or on the ground to remove as much dirt as possible. Wash the filter in gasoline, using a brush to remove dirt from the steel wool. After the filter has been washed, place it face down on two supports. Allow it to drain and dry thoroughly before lubricating.

Lubricate (L). Lubricate or recharge the filter element by dipping it in a bath of oil. In temperatures about 20 F., use SAE-10 oil. Allow the filter to drain thoroughly, intake side down, before it is put into use. While the filter is draining, keep the filter away from places where sand or dirt is being blown through the air. Always replace a filter with its intake side facing the incoming air flow.

Cabinets

The cabinets which house the various components of the set are generally constructed of sheet steel.

Inspect (I). The outside and inside of each cabinet must be inspected. Check the door hinges (if any), the ventilator mountings, the panel screws, and the zero-setting of the meters. Examine the pilot light covers for cracks and breaks. Occasionally remove the covers and see whether the pilot light bulbs are secure in their sockets. Inspect the control panels for loose knobs and switches.

Adjust (A). Adjust the zero-setting of meters if found to be incorrect. Follow the specific instructions given below.

Clean (C). Clean each cabinet including the control panel, outside and in, with a clean dry cloth. Clean the meter glasses and control knobs with a clean dry cloth.

Lubricate (L). Door hinges and latches need little lubrication, but if inspection reveals that they are becoming dry, apply a small amount of instrument oil. All excess oil should be removed with a clean dry cloth.

Meters

Meters are extremely delicate instruments and must be handled very carefully. They require very little maintenance, but, because they are precision instruments, they cannot be repaired in the field. A damaged meter should be replaced with a spare; a defective meter returned to the maker for repair.

Inspect (I). Inspect the leads and connections to the meter. Check for loose, dirty, and corroded connections. Also check for cracked or broken cases and meter glasses. Since the movement of a meter is extremely delicate, its accuracy is seriously affected if the case or glass is broken, and dirt and water filter through. If the climate is damp, it is only a matter of time until enough moisture seeps through a crack to ruin the meter movement.

Tighten (T). Tighten all loose connections and screws. Any loose meter wires should be inspected for dirt or corrosion before they are tightened. The tightening of meter connections requires a special technique because careless handling can easily crack the meter case. To prevent breakage, firmly hold the hexagonal nuts beneath the connecting lugs while the outside nut is being tightened. This permits the tightening of the connection without increasing the pressure of the head of the stud against the inside of the meter case.

Clean (C). Meter cases are usually made of hard highly polished Bakelite, and can be cleaned with a dry cloth. If cleaning is difficult, the cloth should be dampened with cleaning fluid. Dirty connections may be cleaned with a small stiff brush dipped in the cleaning fluid or with a small piece of cloth dipped in the solvent. It should be emphasized that solvents do not remove dirt entirely from hard surfaces. Some of the dirt remains in a softened state and must be removed with a damp cloth. Corroded connections are cleaned by sanding them lightly with a very fine grade of sandpaper, such as #0000. After they are cleaned, the connections should be wiped carefully with a clean cloth.

Adjust (A). Normally, all meters should indicate zero when the equipment is turned off. The procedure for setting a meter to zero is not difficult. The tool required is a thin-blade screw driver. Before deciding that a meter needs adjusting, tap the meter case lightly with the tip of one finger. This helps the needle overcome the slight friction that sometimes exists at the pointer bearings and prevents an otherwise normal unit from coming to rest at zero. If adjustment is needed, insert the tip of the screw driver in the slotted screwhead located below the meter glass and slowly turn the adjusting screw until the pointer rests at zero. Observe following precautions: View the meter face and pointer full on and not from either side. Avoid turning the zero-adjust screw too far, as the meter pointer may be bent against the stop peg or the spring may be damaged. Zero adjustments should not be made for several minutes after shutdown.

Pilot Lights

Pilot lights are used to indicate that power has been applied to a circuit or that a circuit is ready for power to be applied. They are easily removed and replaced. The colored pilot light covers should be removed carefully, lest they be dropped and broken. The maintenance of pilot lights presents no special difficulty, but the following instructions are given for general guidance.

Inspect (I). Inspect the pilot light assembly for broken or cracked pilot light shields; loose bulbs; bulbs with loose bases; loose mounting screws; and loose, dirty, or corroded connections.

Tighten (T). Tighten all mounting screws, and resolder any loose connections. If the connections are dirty or corroded, they should be cleaned before they are soldered. Loose bulbs should be screwed tightly into their bases. Broken or cracked pilot light shields may sometimes be temporarily repaired by joining the broken or cracked pieces with a narrow piece of friction tape. Replace them as soon as possible; also replace broken or burned-out pilot light bulbs as soon as possible. While the removal of a bulb may sometimes be difficult, the process is made simple by folding a small piece of friction tape over the top of the bulb and pressing firmly from the two sides. After the tape is attached, the bulb can be unscrewed and removed from the socket. The socket connections are, of course, inspected while the bulb is out. A new bulb can be replaced with the fingers, but if difficulty is experienced, use friction tape to grip the glass envelope of the bulb.

Clean (C). The pilot light shield, the base assembly, and the glass envelope of the light bulb should be cleaned with a clean dry cloth. Clean accumulated dust or dirt from the interior of the socket base with a small brush. Corroded socket contacts or connections can be cleaned with a piece of cloth or a brush dipped in cleaning fluid. The surfaces are then polished with a dry cloth. Clean contacts and connections are important.

Plugs and Receptacles

There are two main types of plugs and receptacles used to interconnect the various components. The first type of plug is used with a coaxial line and consists of a metal shell with a single pin in the center insulated from the shell. When the plug is inserted into the receptacle, this pin is gripped firmly by a spring connector. There is a knurled metal ring around the plug which is screwed onto the corresponding threads on the receptacle; while the female part is in the plug. The insulation in these plugs is much heavier in order to withstand the voltage. The second type of plug is used for connecting multiconductor cables. The plug usually consists of a number of pins insulated from the shell which are inserted into a corresponding number of female connectors in the receptacle, although in some cases the plug has the female connectors in it and the male connectors are in the receptacle. This type of plug usually has two small pins or buttons which are mounted on a spring inside the shell and protrude through the shell. When the shell is properly oriented and placed in the receptacle, one of these pins springs up through a hole in the receptacle, firmly locking the plug and receptacle together. When it becomes necessary to remove the plug, the other pin is simply depressed and the plug removed. Connections between all plugs and their cables are made inside the plug shell. The cable conductor may either be soldered to the pin or there may be a screw holding the wire to the pin. Remove the shell if it is necessary to get at these connections for repair or inspection. Loosen the screws if there is a clamp holding the cable to the shell. In some cases, it is found that the shell and plug body are both threaded; then the shell may simply be unscrewed. Usually there are several screws holding the shell. These are removed and the shell is pulled off.

Inspect (I). (1) The part of the cable that was inside the shell for dirt and cracked or burned insulation.

(2) The conductor or conductors and their connection to the pins for broken wires; bad insulation; and for dirty, corroded, broken, or loose connections.

(3) The male or female connectors in the plug for looseness in the insulation, damage, and for dirt or corrosion.

(4) The plug body for damage to the insulation and for dirt or corrosion.

(5) The shell for damage such as dents or cracks and for dirt or corrosion.

(6) The receptacle for damaged or corroded connectors, cracked insulation, and proper electrical connection between the connectors and the leads.

Tighten (T). (1) Any looseness of the connectors in the insulation, if possible; if not, replace the plug.

(2) Any loose electrical connections. Resolder if necessary.

Clean (C). (1) The cable, using a cloth and cleaning fluid.

(2) The connectors and connections using a cloth and cleaning fluid. Use crocus cloth to remove corrosion.

(3) The plug body and shell using a cloth and cleaning fluid, and crocus cloth to remove corrosion.

(4) The receptacle with a cloth and cleaning fluid if necessary. Corrosion should be removed with crocus cloth.

Adjust (A). The connectors for proper contact if they are of the spring type.

Lubricate (L). The plug and receptacle with a thin coat of Vaseline if they are difficult to connect or remove. The type of plug with the threaded ring may especially require this.

Chapter 18

CONTROL ROOM AND STUDIO EQUIPMENT

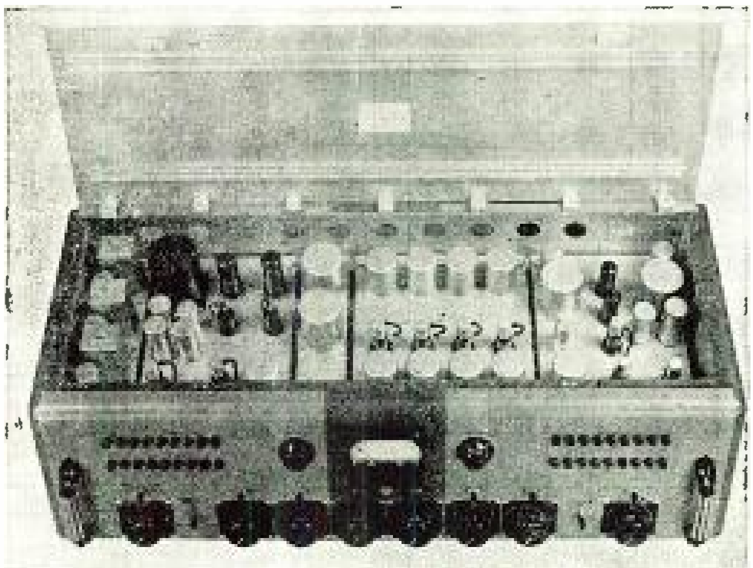
CONTROL-ROOM EQUIPMENT may become very complex in number of circuits and control functions, but is designed and installed to achieve a practical easily operated setup that allows foolproof switching and flexibility of functions. Briefly, the general requirements are as follows:

(a) Amplifiers for stepping up the minute electric energy produced in the microphone by the program sound waves.
(b) Switching and mixing arrangements to allow selection of proper program source and blending of microphone outputs for desired program "balance."
(c) Facilities for "auditioning" or rehearsing a program to follow.
(d) Terminations of inputs and outputs of all amplifiers on jack panels to allow rapid "rerouting" of the signal in case of trouble in any one amplifier or channel.
(e) Incoming and outgoing line terminations on jack panels to permit flexibility in receiving or transmitting the signal in any way desired.

Fig. 18-1 illustrates one type of commercial control-room console which contains all amplifiers and relays within the cabinet. The power supply comes in an external wall mounting unit. This console provides amplifiers, control circuits, and monitoring equipment necessary to handle two studios, announce booth microphone, two transcription turntables, control-room announce microphone, and six remote lines. In addition to this, means are provided for simultaneously auditioning or broadcasting from any combination of studios, turntables, or remote lines. The volume indicator is a standard vu meter which has an adjustable attenuator mounted on the panel to the right of the instrument allowing a 100% deflection of the pointer on the scale to be calibrated for +4, +8, +12, and +16 vu.

The technical layout of this speech input equipment is as follows:

Four preamplifiers connected to four of the six mixers on the panel in center position serve to amplify the outputs of the microphones. A 3-position key switch is in the input of the fourth preamplifier to allow its operation from a microphone in the studio, announce booth,

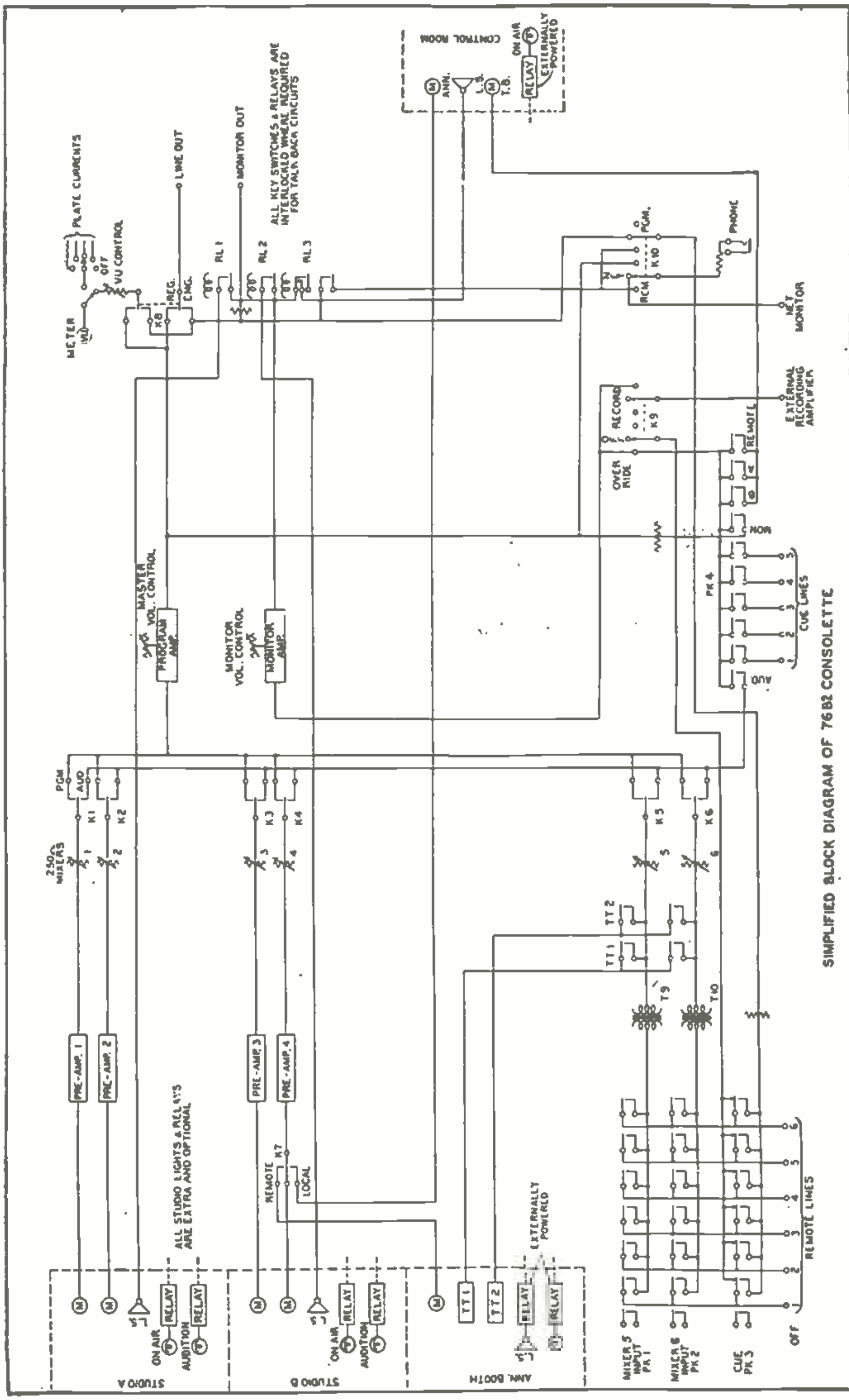


RCA Photo

Fig. 18-1. One type of commercial control-room console containing all amplifiers and relays within the cabinet.

or control room. The outputs of the mixers connect to lever keys to provide switching to the regular program amplifier for broadcasting or to the monitor amplifier for auditioning. When these key switches are operated they also serve to disconnect the studio loudspeaker to prevent feedback, and operate "on-air" light relays. The fifth and sixth mixers may be connected by means of push keys to any of six remote lines or the two turntables. Other push keys on the panel provide circuits for feeding the cue to remote lines and for bringing in monitoring circuits such as transmitter or master-control (where used) outputs. The monitoring amplifier may be used for the program amplifier in emergencies by operating the proper key. Means are also provided to supply power to the preamplifiers from the monitoring amplifier in case of power supply failure to the preamplifiers.

This is an example of the extreme flexibility and emergency provisions designed into control-room equipment. Fig. 18-2 is a simplified schematic diagram of a typical installation. The "Override-Record" switch permits a remote operator to call in from any of the six remote lines and override the program on the control-room speaker. The

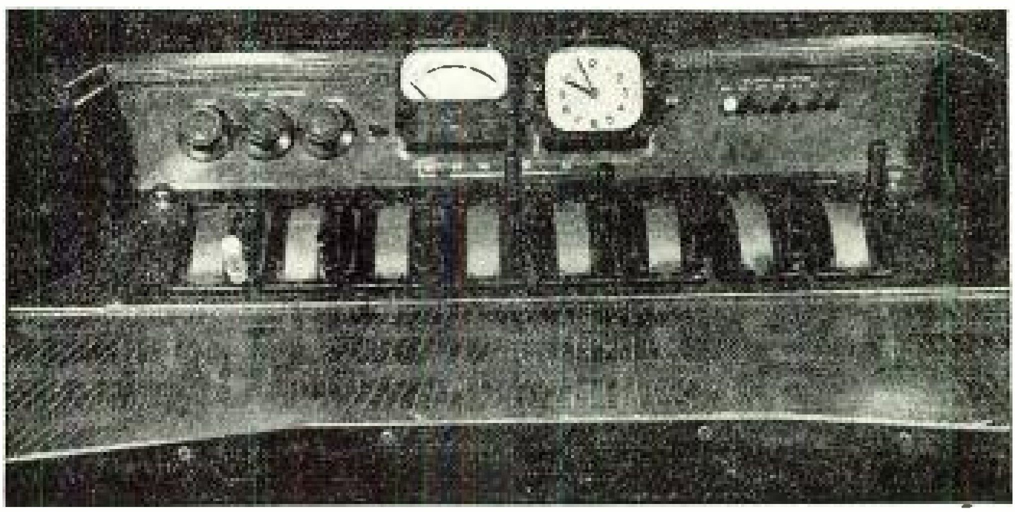


Courtesy RCA

Fig. 18-2. Simplified block diagram of a control-room installation.

"Record" position of this switch furnishes a signal source for an external recording amplifier or other destination.

Although rotary-type faders such as illustrated in Fig. 18-1 are used in the great majority of microphone controls, another type known as the vertical fader or vertical attenuator is gradually coming into wider use than heretofore. A control console using such an arrangement is illustrated in Fig. 18-3. Wherever space permits, there are some advantages to such an installation over the rotary type. It is possible after a little practice to handle two faders simultaneously with each hand, thus doubling the control technique facilities for a large show. The position of this type fader is also more apparent at a quick glance, which is an advantage on complicated productions where the operators attention must be divided between activity in the studio, the script, the producer, and the control console.

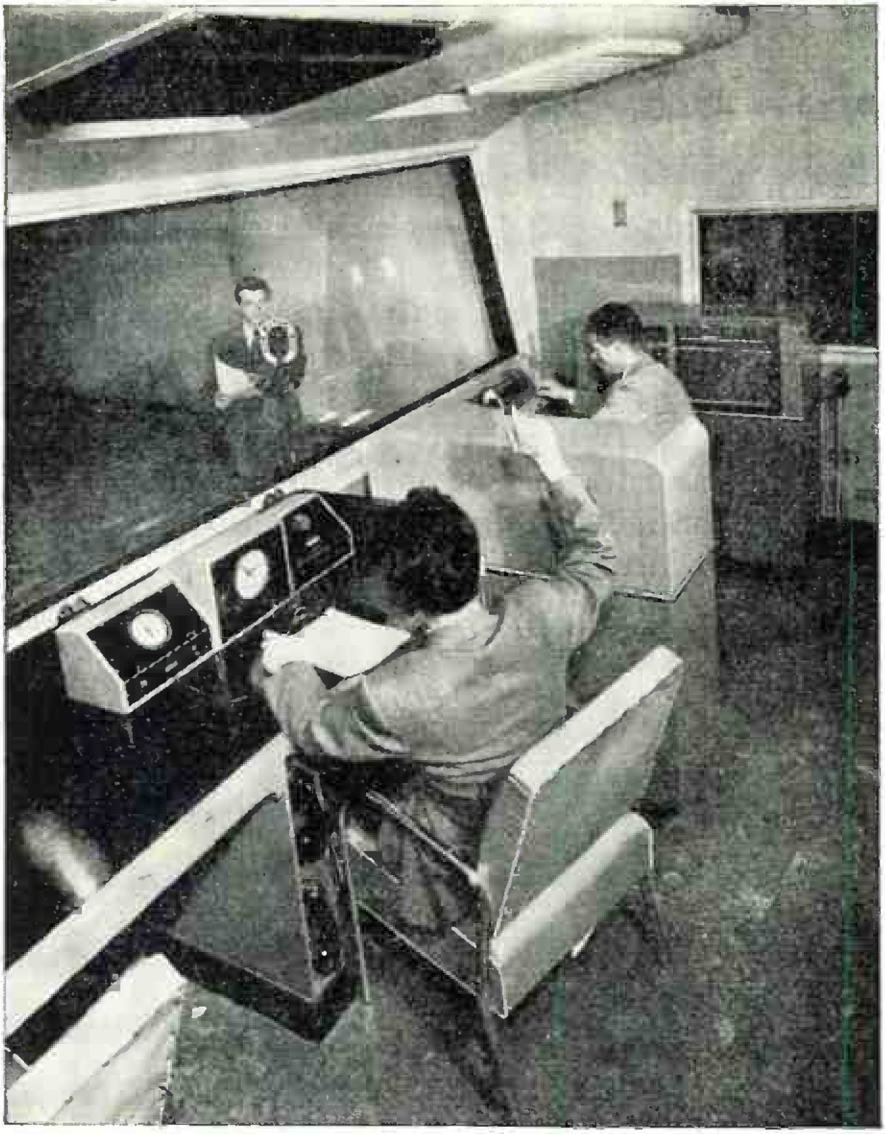


Courtesy Station WHK

Fig. 18-3. The handles of the faders in the control console shown here are vertical, making them easier to handle than rotary faders.

Network centers and large independent stations often use a facility known as the producer's console in conjunction with the individual studio control board. Such an installation is illustrated in Fig. 18-4. The producer's panel is shown at the left of the control console, and contains a talk-back, production timing facilities, and appropriate signaling lights.

Fig. 18-5 shows a central control in combination with a switching system to feed three separate outgoing channels from any of five individual studio consolettes. This switching system consists of the central control and monitoring unit, the master console (both shown in



Courtesy Western Electric

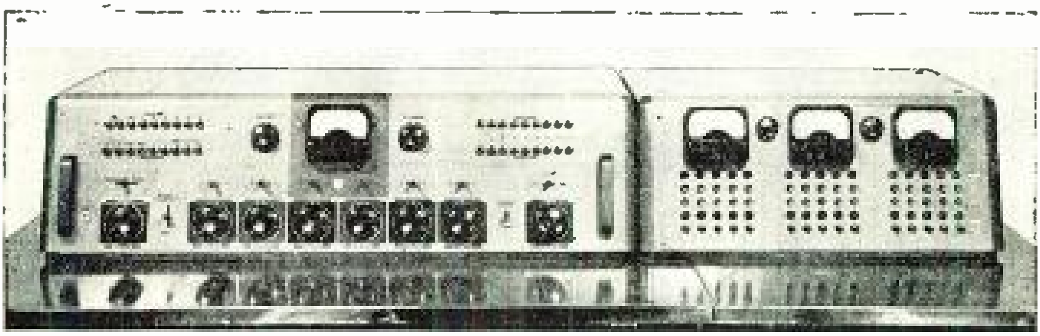
Fig. 18-4. Producer's console is shown on the left with its talk-back and production timing controls. The control console is shown on the right.

Fig. 18-5), plus the individual studio consoles and a remote-control unit (in each studio control) for the master console. The system accommodates one to five studio control consoles.

In a typical system of this kind, the master switcher console is located in the master control room, and one studio switcher installed alongside the consolette in each studio control room. Each studio control is linked to the master control room by two separate cables; the audio cable and the relay control cable. The audio cable connects the studio consolette output terminals to contacts of three relays in the master switcher; the relay control cable connects push-button switches and supervisory lamps in the individual studio switcher (through turnkey switches in the master) to the d-c control circuits of the three relays. Each relay permits connection of the consolette to one of three outgoing channels. The turnkey switches permit the master control operator to assign control of any channel to any studio console. The push-button switches and lamps in the master switcher are connected to the control circuits of all relays, thus allowing switching to be performed either by the master operator or studio operator.

A block diagram illustrating the audio circuits of a typical system, including required accessory equipment, is shown in Fig. 18-6.

A large type of master control with facilities to feed six outgoing channels simultaneously (and more by bridging circuits) is shown in Fig. 18-7.



Courtesy RCA

Fig. 18-5. Master console, and central control and monitoring units for a central control and switching system for feeding three outgoing channels from five studio control consoles.

BROADCAST MICROPHONES

It was only natural that the first devices used for converting sound waves to electrical impulses for sound work and all branches of radio be borrowed from the principle of the telephone. The first so-called "microphone" simply consisted of a telephone type mouthpiece and carbon unit with an appearance like that of Fig. 18-8(A). At just about this same time (around 1921) a microphone known as a "dish-pan" or "phonotron" was developed and used in broadcasting. It consisted of a paper diaphragm somewhat resembling the cone of a loudspeaker, and worked on a magnetic armature principle, illustrated in Fig. 18-8(B). Then came the more familiar carbon microphone (Fig. 18-8(C)) using two carbon buttons in push-pull which resulted in greater fidelity over the single-button telephone-type transmitter. These early carbon microphones, however, not only had a high noise

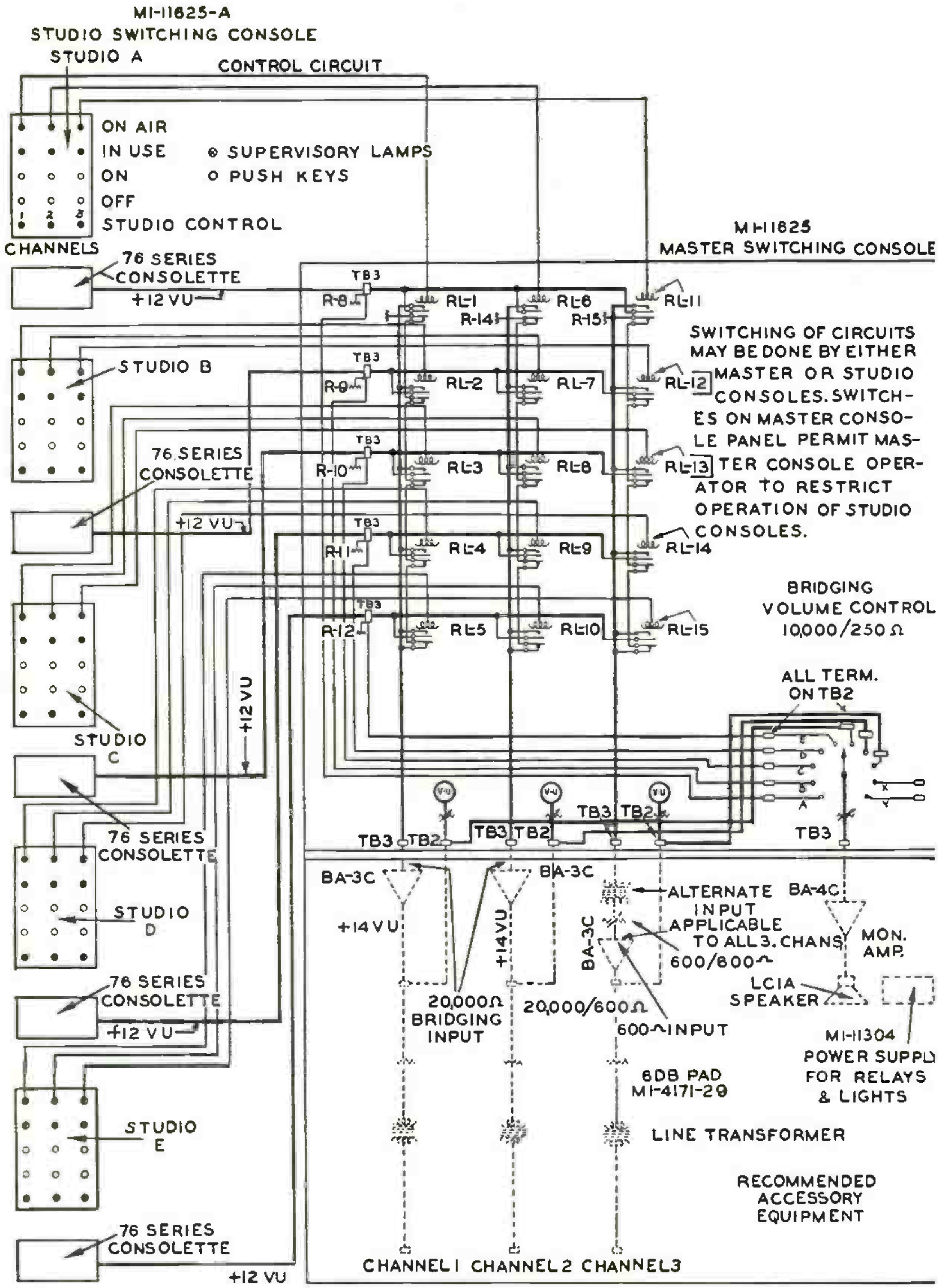
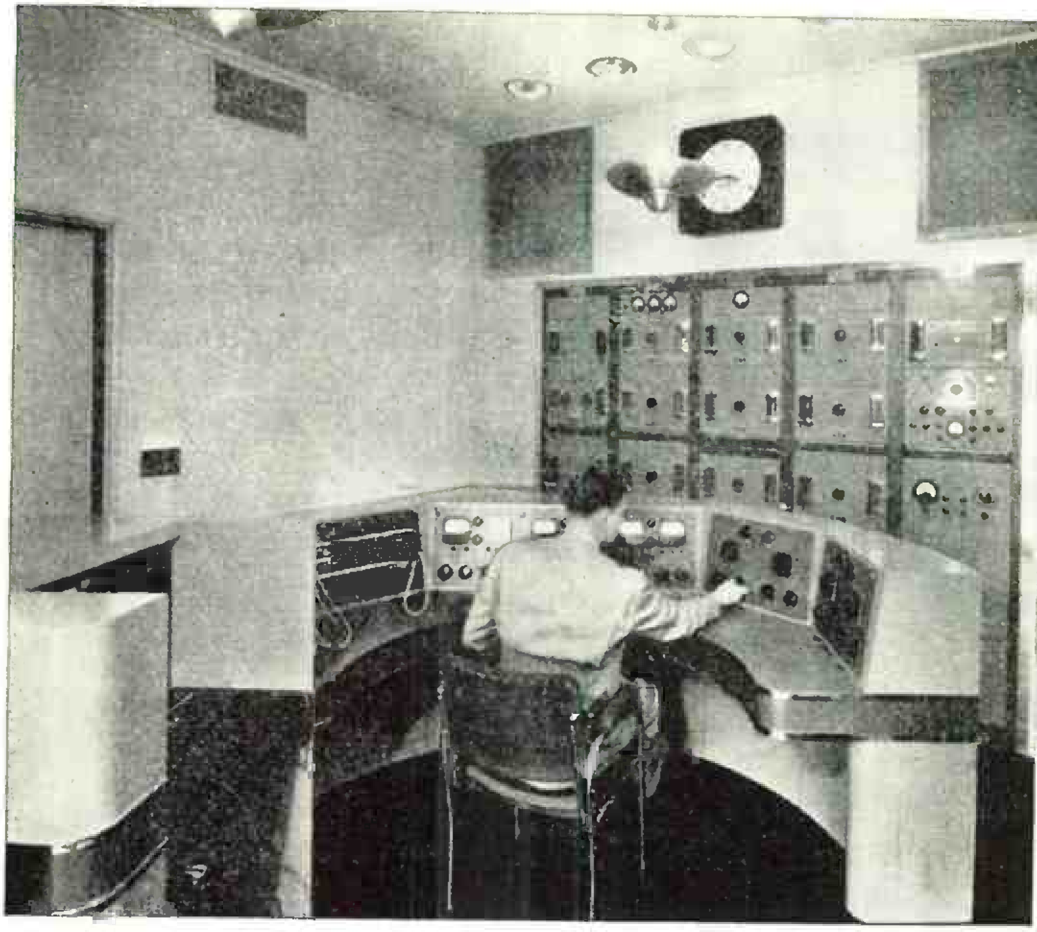


Fig. 18-6. A typical central control and switching system, including accessory equipment.



Courtesy RCA

Fig. 18-7. Master control installation for station with six or more outgoing channels.

level, but were constantly causing trouble due to the cohering of the tiny granules of carbon, resulting in extremely high distortion. These microphones required frequent servicing and "banging" to free the granules sufficiently to permit proper operation.

The first really great advance in microphone design came in 1928 with the condenser microphone. This was termed the "Bullet" mike (Fig. 18-8(D)) and proved of better fidelity and was much more trouble-free than the early carbon units. The condenser principle was shortly followed by the use of crystal elements in microphones, and advancement in microphone design really began to show superior results.

In 1933, the earlier "pressure" idea in microphone design with all the inherent faults of such design was surpassed by the "pressure-gradient" principle, used in the velocity microphone. See Fig. 18-8(E). This microphone, along with the inductor microphone, stands today as the ultimate thus far in sound conversion apparatus, especially for high fidelity use in broadcasting and recording studios and PA installations.

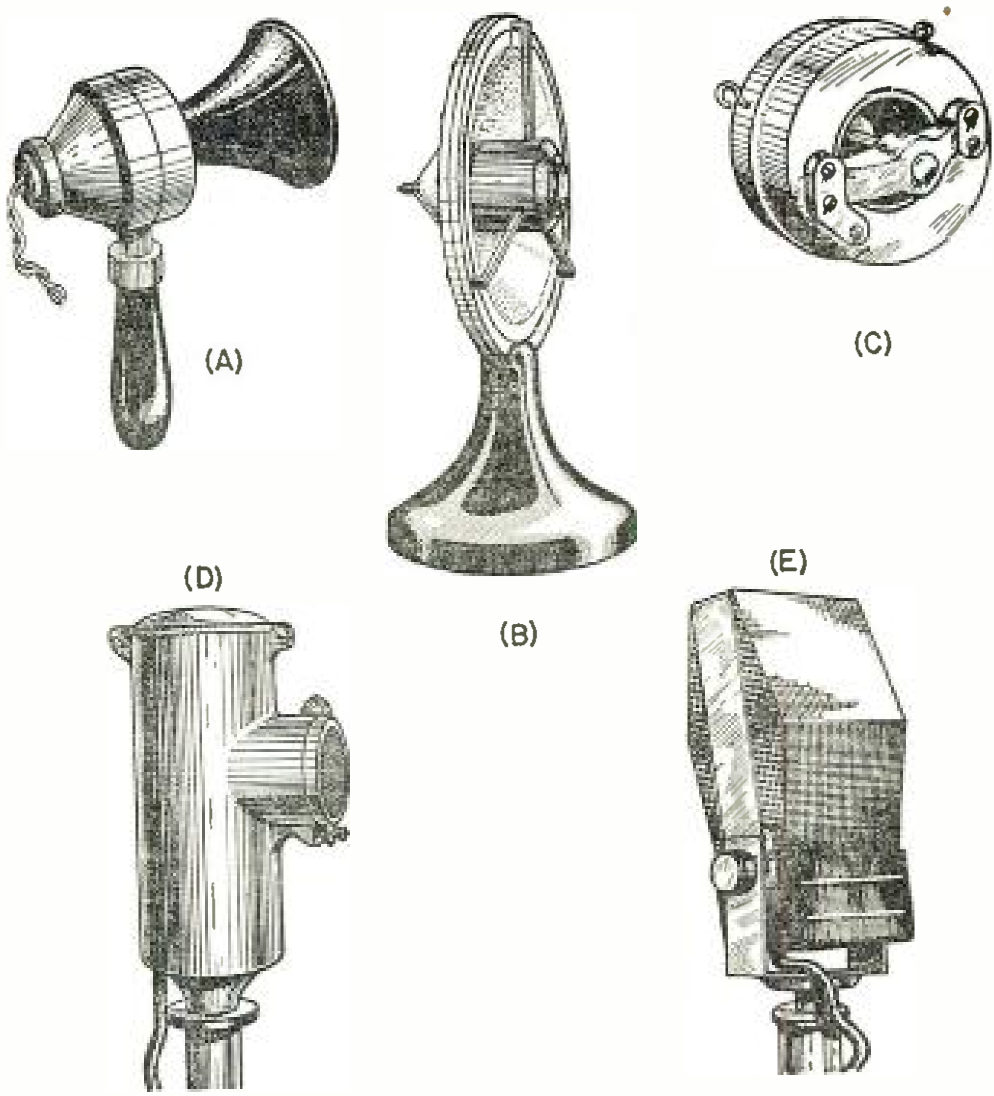


Fig. 18-8. Five types of microphones showing the stages of microphone design from the earliest handphones (A), to the common type in use today (E). (C) is the carbon-button phone which until recently was the standard unit used in microphones.

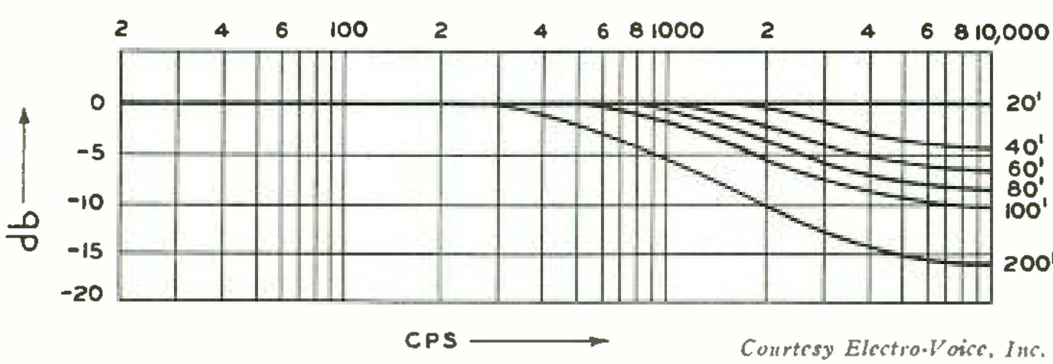
Impedance Value and Cable Lengths

Just as there is need for a wide variety of response patterns, so also is there a need for a great variety of design for permanent or portable use, and a number of impedance values for use in the complex fields of radio and sound.

The output impedance, therefore, of any particular microphone will either be of high impedance that may work directly into the grid circuit of the input amplifier stage, or may use a transformer (30, 150, or 250 ohms) that works into the input transformer of a high-quality amplifier. Some models of microphone that are built for flexible adaptation to various applications have a tapped output transformer with an adjustable screw so that either high- or low-impedance output may be obtained as will be described later, but these are in the minority.

In general, it may be stated that for general communications, amateur radio, some PA installations and some home recorders, a high-impedance microphone is applicable. This is especially true if the microphone cable length may be kept under 25 feet. All high-quality installations use a low-impedance microphone.

The effect of microphone cable length is directly related to the impedance used and the application of the sound system. The metallic shielding of a cable forms a capacitance to the conducting wires which is added to the capacitive effect between the wires themselves. This is equivalent to shunting a capacitor across the line, and thus it becomes obvious that the higher the impedance value used, the greater will be the shunting effect across the line with consequent loss of high-frequency response. Cable length, therefore, must definitely be considered when using high-impedance microphones. The effect of any length of cable is negligible for low-impedance values.



Courtesy Electro-Voice, Inc.

Fig. 18-9. Graph showing the db loss for various microphone cable lengths, at various frequencies. The 20-foot cable is taken as the standard for comparison.

Fig. 18-9 shows the effect of extra cable length over a 20-foot cable. This is based on a "good" microphone cable, with "low" capacitive value of 0.007 μf per 20-foot length. It may, for example, be observed that for a cable length of 200 feet, the frequency response at 5,000 cps would be 15 db lower than for a cable length of 20 feet. It may be realized here why broadcasting stations, recording studios, and high-quality PA installations where long cable lengths are a necessity always use low-impedance microphones. Low impedance must also be used in any form of communication where the operator is working at a point remote from the microphone amplifier requiring long cables, since medium and high frequencies are very important for crispness and intelligibility of speech.

Output and Noise Levels

It is well known that every electrical amplifier has a noise level that depends largely on the thermal noise generated in the grid circuit of the input stage, and is larger as the bandwidth is increased. There is also a noise level in microphones, even though no voltage need be applied to any of the elements. There is, of course, a thermal noise generated in such microphones as the carbon and some condenser varieties where a voltage is incorporated in the microphone elements. But there is also a noise level in all other types due to certain fluctuations in the air pressure itself from thermal velocities of air molecules. Just as the noise level of a particular amplifier will limit the lowest signal volume to be amplified, so the thermal noise in air places a lower limit upon the atmosphere as a useful sound energy medium. In fact, the noise component of the air acting upon the diaphragm of an extremely sensitive microphone is about equal to the thermal noise of a high-quality amplifier. It should be understood, however, that this noise is just about equal to the threshold of hearing of the most sensitive ears between 1,000 and 5,000 cps, and should any microphone (other than a carbon type) noticeably raise the noise level at the output of a good amplifier, some fault will exist within the microphone, cable or input circuit of the amplifier, unless a stray a-c field is being picked up.

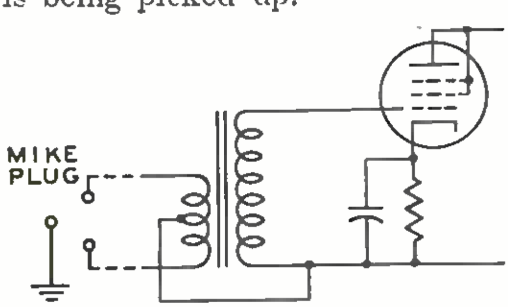


Fig. 18-10. The high-quality low-impedance microphone must be fed to an unloaded circuit to prevent mismatches in the microphone line and reflected resistance loads. The circuit shown here is typical of the type of output circuit used for low-impedance microphones.

High-impedance microphones and carbon microphones intended for general communications have a somewhat higher output level than low-impedance, high-quality units. The output level ratings of low-impedance microphones are always given as an open circuit rating or unloaded transformer input rating. It is quite important in most low-impedance installations that the microphone be worked into an unloaded grid circuit. See Fig. 18-10. High-quality mikes such as the ribbon and combination types may have their response characteristics seriously affected by a reflected resistance load on the mechanical constants of the moving elements.

When a microphone is working into an open circuit of this kind, the output cannot be expressed in terms of db power level since no measurable power is expended in an open circuit termination. These microphone ratings are usually given as "effective output level," and is so calculated that when a known amplifier gain is added to the output level of the mike in db, the measured output level of the amplifier is obtained.

Volume Handling Capability

Just as there is a minimum level of sound waves that can be adequately picked up by a microphone, so also is there a maximum sound pressure that may be handled without distortion. This is particularly true of any pressure-type unit. Remember that it is the function of the diaphragm which is connected to the moving element inside to move physically in accordance with the applied sound pressure, thus providing a voltage image of the sound. There is, however, an elastic limit to this movement. Suppose the diaphragm is connected to a moving coil as in the inductor (also called dynamic) microphone. Study Fig. 18-11(A) for a moment. This is what happens when an extremely loud sound pressure exists, such as shouting at close range into the unit. At point *a*, the sound is still increasing, but the elastic limit of the diaphragm has been reached, and it has come to rest.

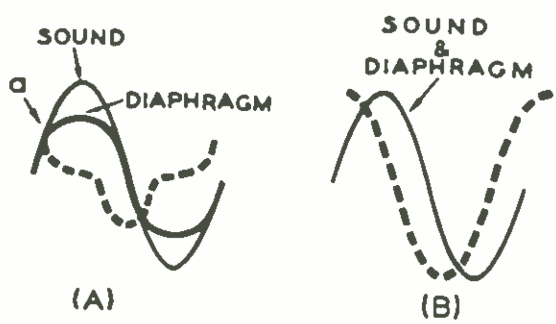


Fig. 18-11. (A) and (B) show the movement of the diaphragm of a dynamic-type microphone under various sound pressures. In (A) the diaphragm does not follow the sound wave identically because the pressure of the sound wave is too excessive. (B) shows the action of normally loaded microphone.

This means that the coil has also come to rest, and the voltage output has dropped to zero, even though the air pressure is still increasing on the diaphragm. The diaphragm then returns at high velocity as the pressure subsides, a surge in voltage is created and the distortion is shown by the dotted voltage line. This kind of waveshape results in "blasting" and contains a number of distortion components and false harmonics. Fig. 18-11(B) shows how the voltage is generated in a normal microphone that is not overloaded, and follows the shape of the sound pressure in exact form.

Output Level Ratings

Although the output level of a microphone is nearly always given in terms of db, it is highly important to avoid confusion by noting whether the db reference is for power or voltage, and what the "zero db" response level is. In general, low-impedance microphones are rated in terms of power on a basis of 0 db = 1 mw. However, some manufacturers have 0 db = 6 mw or 12.5 mw.

The output level ratings of high-impedance units such as the crystal microphone are given in terms of db of voltage, usually on the basis of 0 db = 1 volt for a sound wave exerting a pressure of 1 dyne/cm^2. In some instances, a sound pressure of 10 dynes/cm^2 is used in the rating. If it is desired to convert from a pressure of 10 dynes/cm^2 to one of 1 dyne/cm^2, it is necessary to subtract 20 db from the given rating. In many cases the output is stated as a certain number of db below 1 volt per bar. A bar is equal to 10^6 dynes/cm^2.

PRESSURE-TYPE MICROPHONES

Condenser Microphones

This type of microphone was the second important development in the evolution of sound-converting apparatus, being the successor to the familiar carbon mike. It is similar in construction to the carbon mike, must have a voltage fed back to the elements, and depends for its operation on variations in capacitance rather than variations in a carbon cup resistance. The output is much lower than the most highly damped carbon unit, but it is inherently much quieter in operation with low noise level. The average condenser type has an extended frequency range over the carbon type. Its decline in use may be laid to the fact that, for portable communication uses, the higher output level of carbon type is useful; and, for other applications requiring lower noise level and extended frequency range, the dynamic, crystal, or velocity unit is much better and requires no associated voltage supply.

In almost every instance it is necessary to have the preamplifier in the same case with the condenser unit. The voltage output of the microphone itself is extremely feeble, and the associated preamplifier preferably uses d-c filaments to avoid any a-c fields causing hum pickup. The output of the preamp is 200 to 500 ohms (using a transformer) so that the signal may be delivered to the main amplifier with negligible loss of high frequencies. A typical condenser microphone circuit is shown in Fig. 18-12(A). Constructional details are illustrated in Fig. 18-12(B).

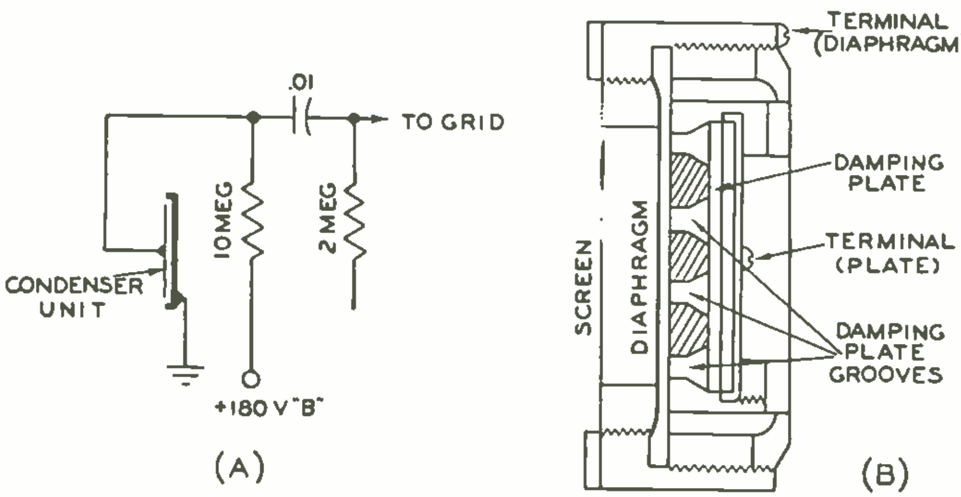


Fig. 18-12. Schematic diagram of a typical condenser microphone (A), and the constructional details for the main unit (B).

As shown in the circuit diagram, the polarizing voltage is fed through a high resistance of the order of 10 megohms. This resistance must be much higher than the reactance of the microphone, if not, the frequency response is seriously affected. It should be apparent to the practical man that the total effective resistance into which the condenser unit works is actually the parallel combination of this resistance and the grid-leak resistance of the input tube.

The diaphragm is of light aluminum alloy with high tensile strength, and serves as a movable electrode of a condenser. The back plate serves as a fixed electrode. The same kind of damping is employed as in the carbon unit.

Another important feature of a pressure microphone which has not thus far been discussed is what is known as cavity resonance. In microphone construction there is a small cavity in front of the diaphragm, (see Fig. 18-12(B)). There is, of course, a fundamental resonant frequency existing in this space depending upon the size. In the average condenser microphone, this cavity increases the pressure on the diaphragm about 5 db at 3,500 cps. This rise in force due to the cavity resonance starts at about 500 cps and attains an ultimate increase of about 6 db at 4,500 cps. The diaphragm is stretched to a tenseness such that its resonant frequency lies between 5,000 and 10,000 cps.

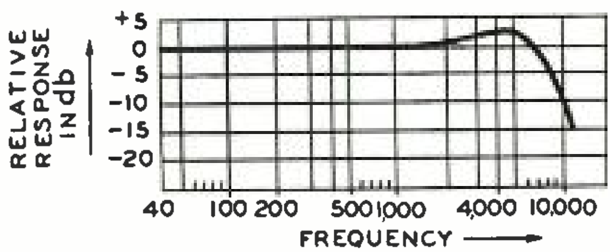


Fig. 18-13. Relative response (in db) for various frequencies for a typical condenser microphone.

The frequency response of a representative type of condenser microphone is shown in Fig. 18-13.

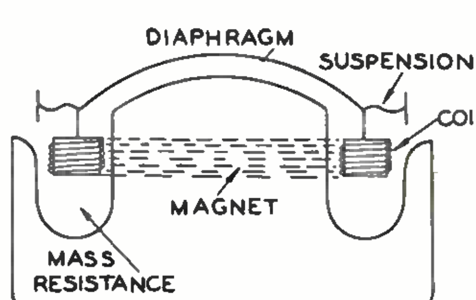
MOVING-CONDUCTOR MICROPHONES

This type of mike is known variously as "moving-conductor," "inductor," or "dynamic" variety. It is one of the most popular current types in all fields of radio and sound system applications. The electrical output results from the motion of a conductor (usually a small coil) in a magnetic field. The magnets are of the permanent type and require no external voltage.

There is actually a slight technical difference between a "moving-coil" microphone and the "inductor" type, although the principle of operations are identical. The moving-coil type, as its name implies, utilizes a small voice coil in the magnetic field. In the inductor type, the diaphragm actuates a straight conductor that is suspended between the magnets.

Fig. 18-14 shows the skeletonized basic construction of the moving-coil (dynamic) unit. Motion of the diaphragm in the sound field causes the small coil to move back and forth in the magnetic field

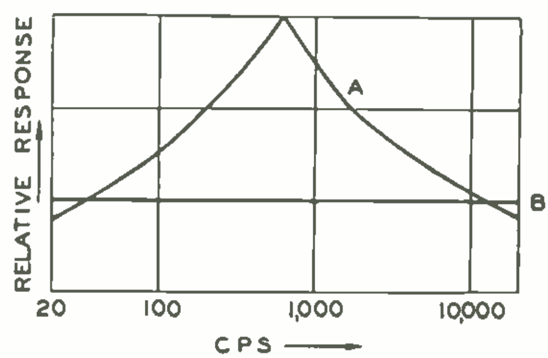
Fig. 18-14. Moving-coil (dynamic) microphone movement. The motion of the diaphragm is transmitted directly to the moving-coil which, when it moves in the magnetic field crossing it, generates and emf.



in accordance with the actuating sound pressures. In practice, this small voltage thus generated is applied across a transformer, the primary of which matches the impedance of the coil. Thus it may be seen that a varying audio voltage is impressed across the secondary of the transformer; this secondary being of an impedance value of 50, 150, 250 ohms, or high impedance depending upon the application desired. The mass resistance on the opposite end from the coil is used to balance the movement of the diaphragm.

However, a diaphragm and coil assembly of this kind, without a sort of resistance control or damping, would have a frequency response somewhat similar to that of Fig. 18-15 curve *a*. Curve *b* is, of course, the theoretical "ideal response curve" of a microphone, and this illustration shows the necessity for some kind of resistance control.

Fig. 18-15. Right. Curves representing the ideal response (B) and the actual frequency response (A) of a dynamic microphone without damping.



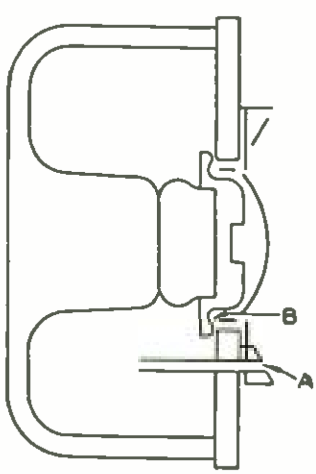
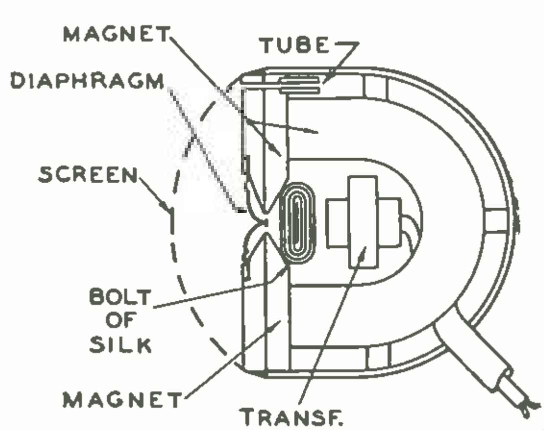


Fig. 18-16. Left. Dynamic-type microphone showing special constructional features which accomplish damping and compensate for mass reactance of the diaphragm at high frequencies.

After Olson, Van Nostrand Co.

Fig. 18-17. Right. Skeletonized version of an inductor-type microphone showing the use of a bolt of silk for damping, as well as an air tube for increasing low-frequency response.



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Damping is accomplished in several ways in this type of instrument. Fig. 18-16 shows the addition of two elements to the construction aside from the case housing the entire unit. These additions are tube *a* and air slits marked *b*. In some microphones, silk cloth is used instead of the air slits shown. Mass reactance of the diaphragm at high frequencies (which reduces high-frequency response) is overcome by these air gaps or by silk cloth, resulting in an increase of over an octave at the high end of the scale. Increase in low-frequency response is obtained by the addition of the tube *a* and its associated air cavity. The addition of these features is designed to maintain uniform diaphragm velocities at all frequencies. Fig. 18-17 shows the basic constructional features of an "inductor" type and its means of damping, illustrating the difference to the "moving-coil" type. Fig. 18-18 is the complete constructional details of a representative type of dynamic microphone.

Most voice coils used in the dynamic microphone are wound with aluminum ribbon or wire. The use of aluminum provides a maximum ratio of conductivity to mass. When ribbon is used, it is found to be wound edgewise on the form.

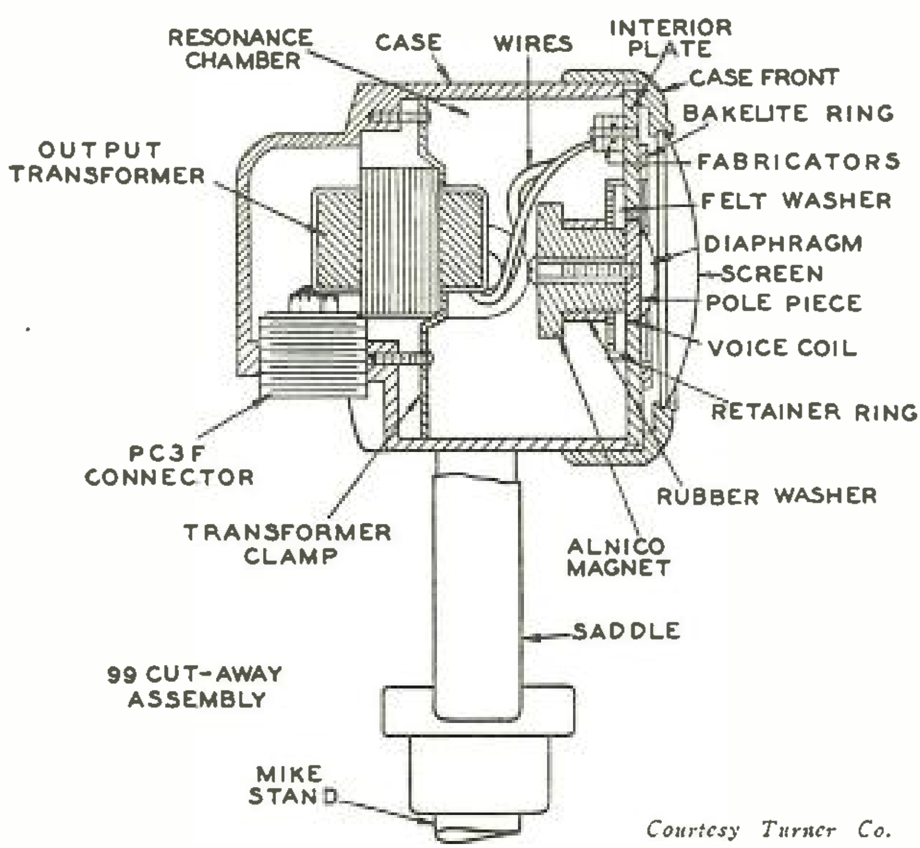


Fig. 18-18. Constructional details of a typical dynamic-type microphone.

Diaphragms of the dynamic and inductor types are found variably made up of either aluminum alloys, styrol, bakelite, or paper.

PRESSURE-GRADIENT MICROPHONES

This microphone is variously known as pressure-gradient, velocity, or more commonly the "ribbon" type. Briefly, it may be described as a moving-conductor microphone in which the conductor is in the form of a ribbon suspended so as to vibrate freely in a magnetic field. This ribbon itself constitutes its own diaphragm, and since it is not housed in any closed type of case and is exposed freely to the air on its two active sides, there are no adverse effects of cavity resonance, diaphragm resonance, or pressure doubling. This type of sound pickup is extremely popular in the broadcasting industry due to the very faithful reproduction obtainable.

Fig. 18-19 illustrates the basic construction of a ribbon microphone. The ribbon consists of a thin corrugated aluminum ribbon (aluminum alloy to minimize mass), and is suspended between the poles of a permanent magnet as shown. Thus any movement of the ribbon caused by passing sound waves will cause the magnetic lines of force to be cut transversely inducing a corresponding voltage between the two ends of the ribbon. In order that the ribbon constitute a mass re-

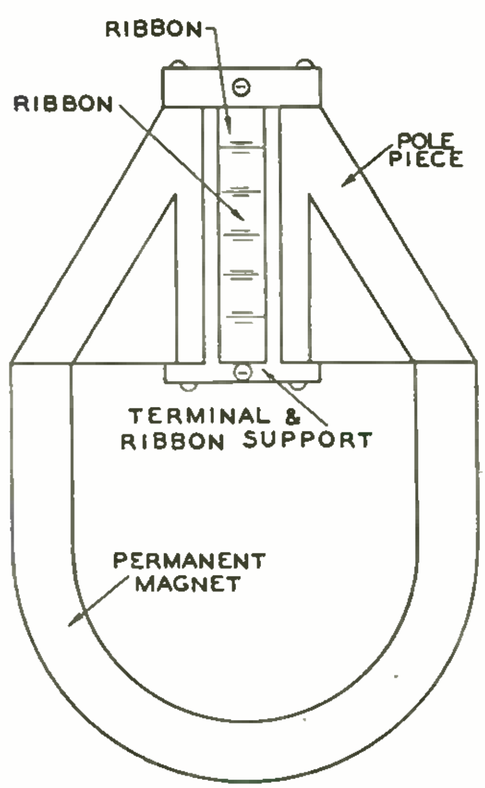
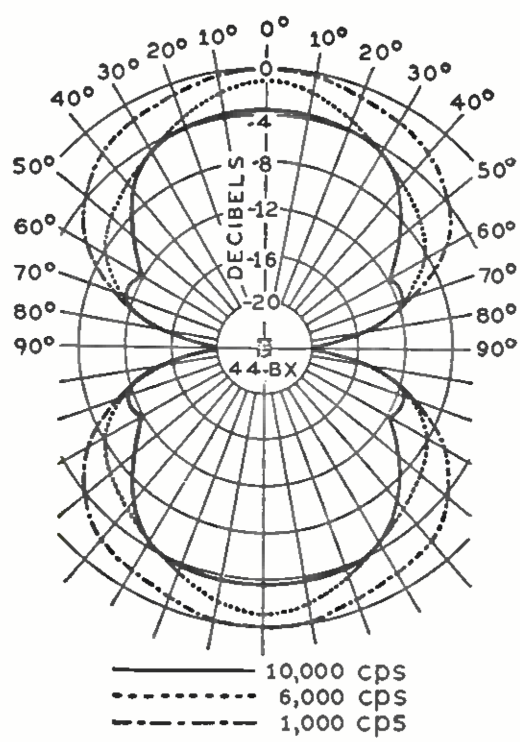


Fig. 18-19. The movement of a ribbon-type microphone. The ribbon (a thin corrugated aluminum strip) is suspended between the pole pieces of a permanent magnet. As is seen from this diagram, the ribbon will be put in motion only by sound waves coming from the front or the rear of the page, resulting in a bidirectional response.

actance over the usable frequency range, its resonant frequency is made lower than the lowest frequency to be reproduced. This eliminates "peaks" at any resonant frequency.

The ribbon moves through the magnetic field as a function of the difference in sound pressure between two points in free space. Since this differential pressure exists between front and back of the ribbon, it will move in the direction of diminishing pressure. Thus, under certain conditions, it is said to correspond in motion to the particle velocity in a sound wave, hence the term velocity microphone. This rate of change of pressure with distance is called "pressure-gradient."

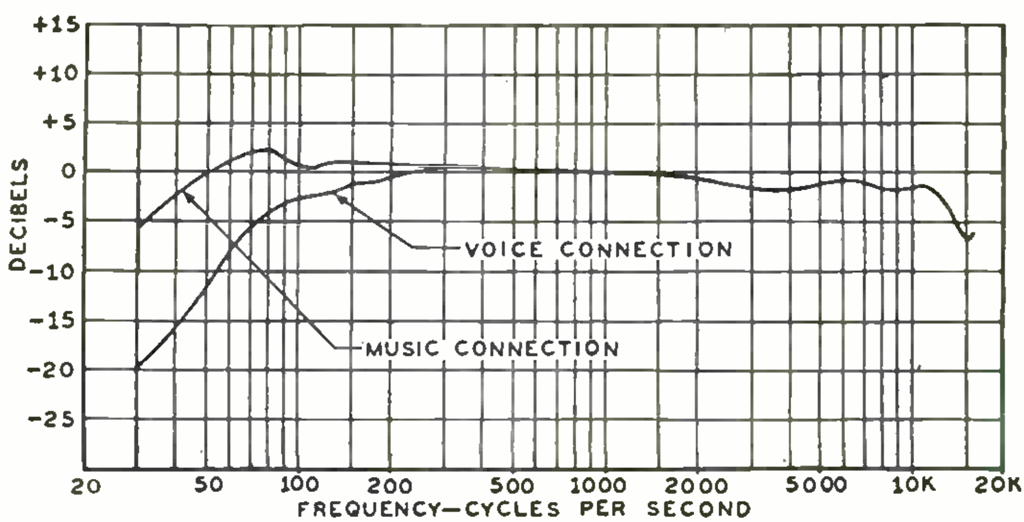
The ribbon can respond only along the axis perpendicular to its surface, and is "live" only from either "face," and dead to sound waves from the side or 90° from zero axis. Hence this microphone is, by the nature of its design and construction, a bidirectional instrument with a polar response pattern.



Courtesy RCA

Fig. 18-20. In (A) is shown an external view of a ribbon-type (velocity) microphone. The family of response curves for the ribbon-type microphone is shown in (B). These curves show how the response of the microphone varies for three frequencies at different angles with the microphone.

It should be understood that although the pressure of a sound wave in air may be independent of frequency, the pressure-gradient, or *rate of change* of pressure with distance is not. Therefore, the mechanical impedance of a ribbon is *proportional to the frequency* of the actuating sound waves. In practice, this is only important where the spherical character of speech waves very close to the mouth will cause a greater pressure on the ribbon at the low frequencies than results from the higher-frequency components of speech. Thus it is commonly noticed that when talking into a ribbon microphone at close range there is a definite "boominess" resulting from an accentuation of the lower frequencies. This is overcome in all modern ribbon instruments by use of a "speech strap" when the microphone is to be used for speech only. It consists of an adjustable tap on the output transformer (which raises the impedance of the ribbon from its extremely low value to 50, 150, or 250 ohms) and lowers the open circuit reactance of the transformer such that low-frequency response is decreased. The tap is adjustable by means of a screw on the bottom side of the microphone case and may be placed in the "music" or "voice" position at will.



Courtesy RCA

Fig. 18-21. The ribbon microphone has a "voice" connection and one for "music" to obtain the best response for either type. The frequency response curves for both types are shown above.

The ribbon microphone is typified by the RCA 44-BX, illustrated in Fig. 18-20(A), along with the polar response pattern at three frequencies, shown in Fig. 18-20(B). This microphone comes with a tapped transformer for 50/250 ohms balanced to ground, with an output level of -55 db referred to one milliwatt and a sound pressure of

10 dynes/cm². The frequency-response curve for both voice and music position of the tapped transformer is illustrated in Fig. 18-21.

UNIDIRECTIONAL MICROPHONES

The fundamental theory of one type of unidirectional microphone was given in Chapter 4. This feature of directivity is a very popular and useful characteristic for discriminating against wanted and unwanted sound. The field of application is endless. The PA engineer must guard against feedback. The communications engineer is interested, generally, only in a voice immediately at one side of the instrument, the same for the ham operator. Recording and broadcasting technicians are concerned with the same general principles of wanted and unwanted sounds, and *all shades in between* for proper balance of various sound sources. Thus the unidirectional microphone is the most popular mike in use today.

As brought out in Chapter 4, the first approach to achieving a truly directional instrument was the combination pressure and pressure-gradient principle. This results in effectively annulling one loop of a bidirectional pattern, making the microphone sensitive only to sounds from one side of the mike. Since the ribbon element is the most practical for a pressure-gradient microphone, and the moving-coil or dynamic principle has proven highly effective (due to light weight and freedom from temperature effects) for the pressure microphone, these elements were used in the unidirectional instrument. A review of this part of Chapter 4 will be helpful here.

A simple connection of the two elements in series is only the beginning, however, of obtaining a practical unidirectional response. This is so because the magnitude of response for a given sound pressure, and the phase of the output voltage differ from one another over the usable frequency range particularly at low and high frequencies. For this reason, certain deviations in design and construction are apparent in the microphone from the conventional ribbon and dynamic units.

Firstly, the diaphragm of the dynamic unit must be as close as possible to the ribbon in order to minimize phase differences resulting from distance of separation. At the same time, there must be no adverse effects of either element by any disturbance of normal operation by the proximity of the units. Thus the dynamic unit housing must be streamlined so that it has minimum effect on the closely neighboring ribbon element, and the ribbon assembly uses a wider than normal air gap requiring heavier permanent magnets.

Secondly, since the two elements have essentially the same magnitude and phase response at middle frequencies but may deviate considerably at low and high frequencies, an electrical equalization or corrective network must be used in the output of the two elements before their combination at the output transformer.

The Western Electric 639A and 639B cardioid microphones are typical representatives of the combination ribbon and dynamic microphones. Fig. 18-22 illustrates the internal construction of this instrument.

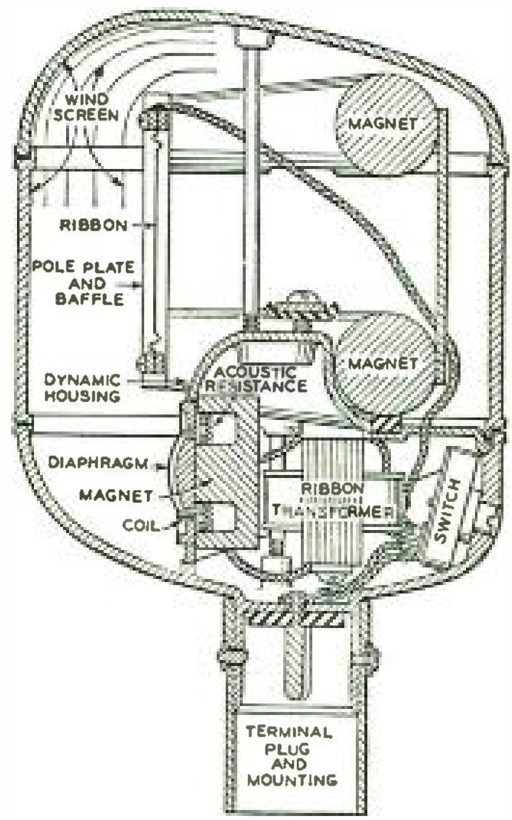


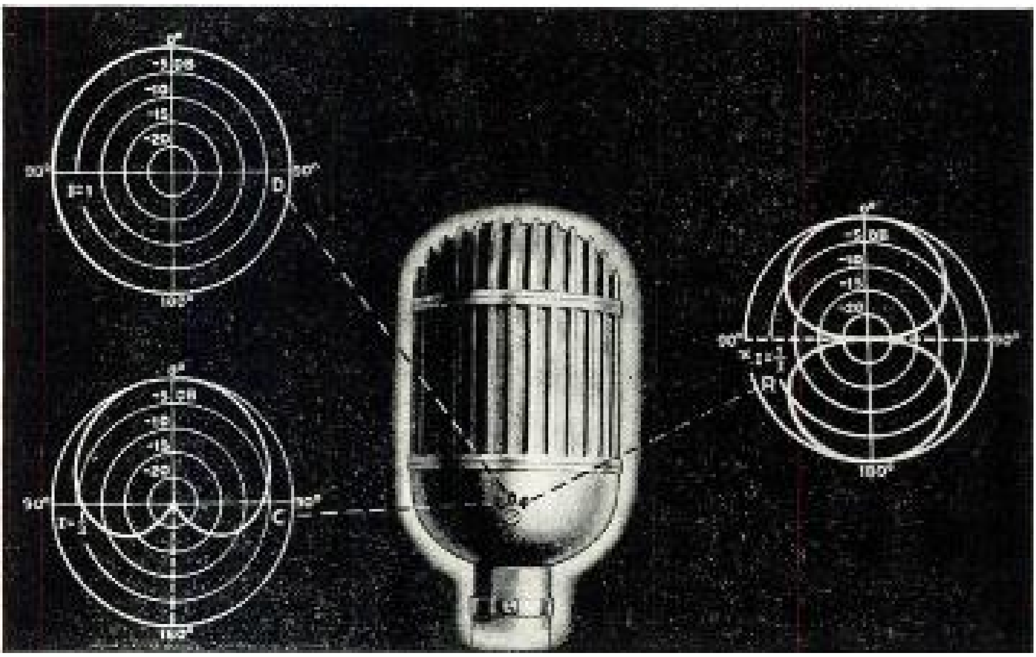
Fig. 18-22. The construction details of the cardioid-type microphone, which provides a number of response patterns by means of a six position switch in the rear of the microphone housing.

Courtesy Western Electric

The ribbon element in this microphone was redesigned to meet the above requirements and also to reduce flutter caused by wind when used outdoors. This ribbon is about half the length of a normal ribbon, is somewhat narrower, but greater in thickness. This thickness results in a stiffness about 10 times that normally employed in a velocity microphone, with greater stability and consequent reduction of noise caused by wind.

The lower part of the microphone case contains the dynamic unit and housing, in which is located the ribbon transformer, electrical equalizer, and variable pattern switch.

This switch permits use of the microphone either as dynamic unit alone, ribbon unit alone, or both in series to achieve the unidirectional response pattern. Fig. 18-23 illustrates the 639A type. With the switch in the *D* (dynamic) position, the response is nondirectional as shown. In the *R* (ribbon) position, the ribbon alone is used and the usual bidirectional pattern is obtained. To obtain the cardioid or unidirectional response, the switch is turned to the *C* position which utilizes both elements through the electrical equalization circuit. The output level for either ribbon or dynamic element alone is 90 db below 1 volt per dyne/cm². The cardioid position yields an output of –84 db.



Courtesy Western Electric

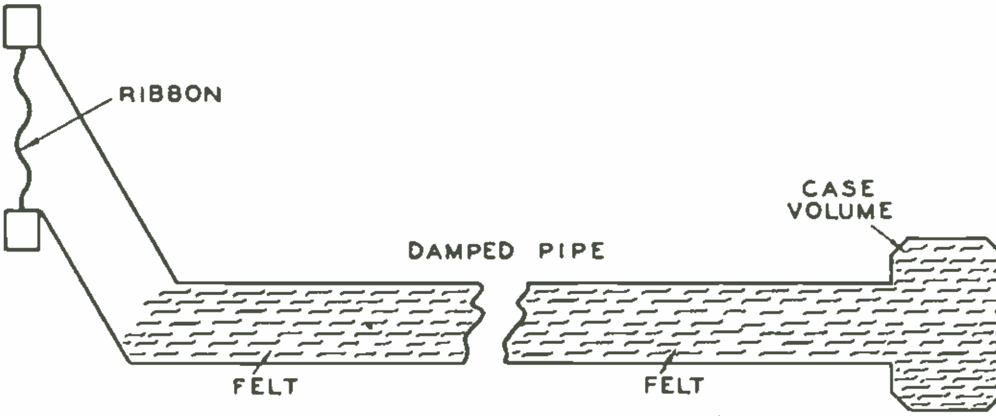
Fig. 18-23. Typical cardioid-type microphone showing the response and directional patterns resulting from the various positions of the switch at the base of the microphone.

The Western Electric 639B is the same microphone with the addition of three extra response patterns as was illustrated in Chapter 4. The output impedance of either microphone is 30 ohms.

Other Methods of Obtaining Unidirectivity

There are several very popular means of constructing a microphone with unidirectional response other than the combination ribbon and moving coil just described. Their basic action, however, is based upon the same general principle of combining a pressure-gradient unit.

Fig. 18-24 illustrates a method of using a ribbon element such as commonly used in velocity microphones to act as a pressure-operated device. It consists of a ribbon suspended in the usual fashion in the



After Olson, Van Nostrand Co.

Fig. 18-24. One method of making a microphone unidirectional is to leave one face of the ribbon freely exposed to sound waves and to back up the other side with an infinite impedance to sound as shown here.

magnetic field of a permanent magnet, but differs from the velocity type in that the "face" is exposed to the atmosphere whereas the rear side is terminated in an acoustic resistance. Thus the back of the element is enclosed such that it presents an infinite impedance to sound, resulting in a microphone that is pressure-operated with non-directional characteristics (except at high frequencies) as described for other types of pressure elements in Chapter 4.

By combining this principle of ribbon action with that of the pressure-gradient ribbon action, a unidirectional response pattern is

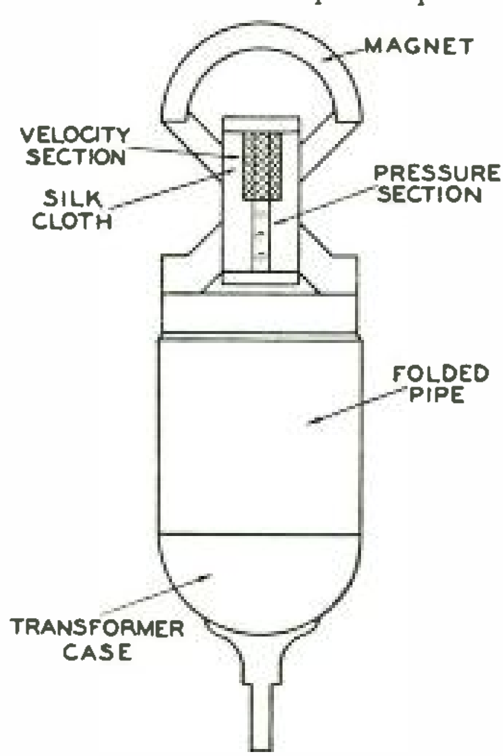


Fig. 18-25. A unidirectional microphone consisting of a ribbon in a magnetic field; the ribbon is velocity actuated at the upper half, and pressure-gradient actuated at the lower half.

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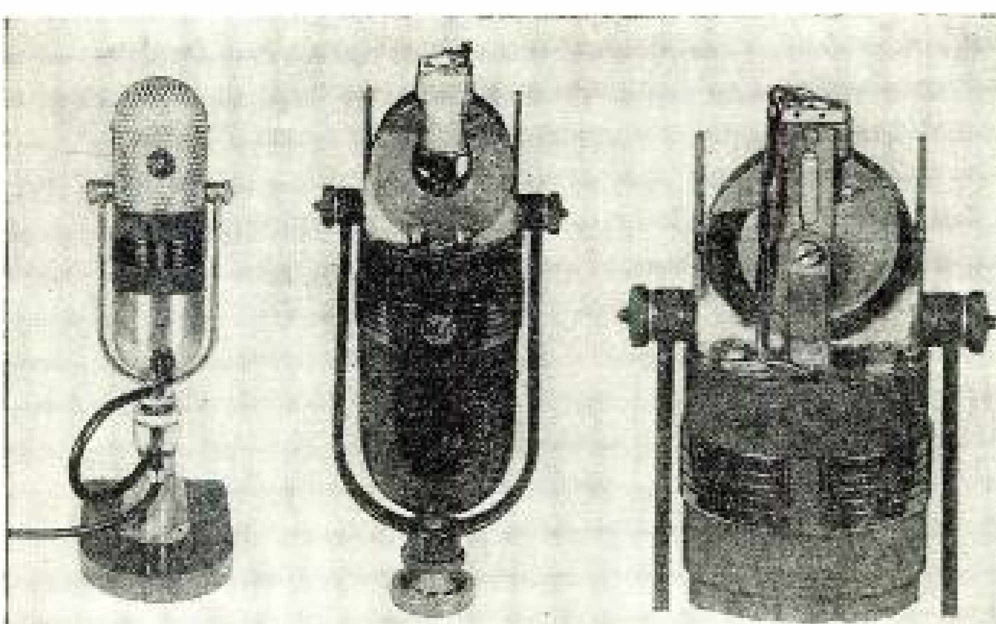
obtained. Fig. 18-25 illustrates this method, where one continuous ribbon is used, with the upper half acting as a velocity (pressure-gradient) microphone, and the lower half acting as a ribbon pressure device terminated in the rear with a folded pipe acting as the acoustic resistance. The pipe is usually damped with tufts of felt.

Due to the length of pipe behind the pressure section, the velocity of the pressure ribbon leads the pressure in the sound waves at the low audio frequencies. To compensate for this characteristic, a cloth screen (shown in Fig. 18-25) is placed in front of the velocity section which introduces a corresponding phase shift in the velocity ribbon. Phase shifts at high frequencies is minimized by using the same ribbon element for both units and suitable geometrical configurations of the field structure.

RCA Type 77-D Variable Pattern Microphone

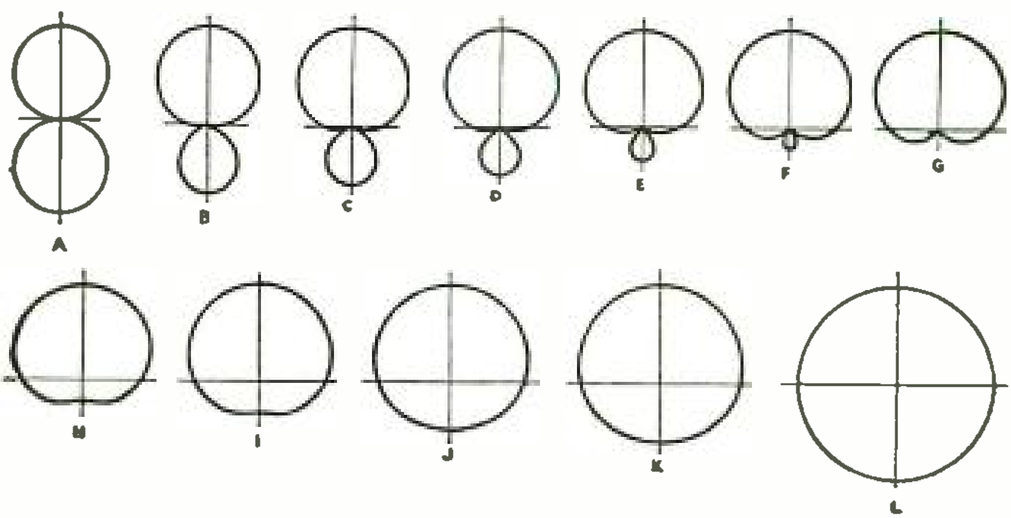
This microphone is of somewhat similar design to the above principle, with the added feature of a means of varying the acoustic impedance presented to the ribbon.



Courtesy RCA

Fig. 18-26. Rear external view of the polydirectional microphone. The middle view is the front of the ribbon assembly and the rear view is at the right.

Fig. 18-26 shows three views of this microphone. Directional characteristics may be changed by varying the area of an adjustable openning in the labyrinth connector. (This labyrinth in the bottom of the case, as that of Fig. 18-25.) When the opening is so large that the back of the ribbon is entirely exposed to the air (as in the ordinary velocity microphone), the acoustic impedance is zero and the response pattern is bidirectional. When this aperture is completely closed, the acoustic impedance is infinite and the microphone becomes nondirectional in accordance with the usual pressure-operated device. By varying the size of the opening, a great variety of response patterns are obtainable. Fig. 18-27 shows the number of patterns obtainable with this microphone. Fig. 18-28 shows the frequency response for the various positions as well as for the associated "speech-music" position of the switch located on the bottom of the case. (M, V_1, and V_2).



Courtesy RCA

Fig. 18-27. Various response patterns obtainable with the polydirectional microphone. The directional characteristics are varied by changing the area of the opening of the damping tube of the pressure-gradient section of the microphone ribbon.

The aperture is adjustable by a slotted shaft on the wind screen which is brought out flush to the designation plate. This plate is marked "U" for unidirectional, "B" for bidirectional and "N" for nondirectional. Three additional markings are used as reference points for other patterns obtainable in this instrument.

Output impedances come in 50, 250, or 600 ohms with three-conductor cable (balanced to ground). Output level is −59 db referred to a reference level of one milliwatt and a sound pressure of 10 dynes/cm². The response curves of unidirectional, bidirectional, and nondirectional positions of the adjustable shaft are shown in Fig. 18-29.

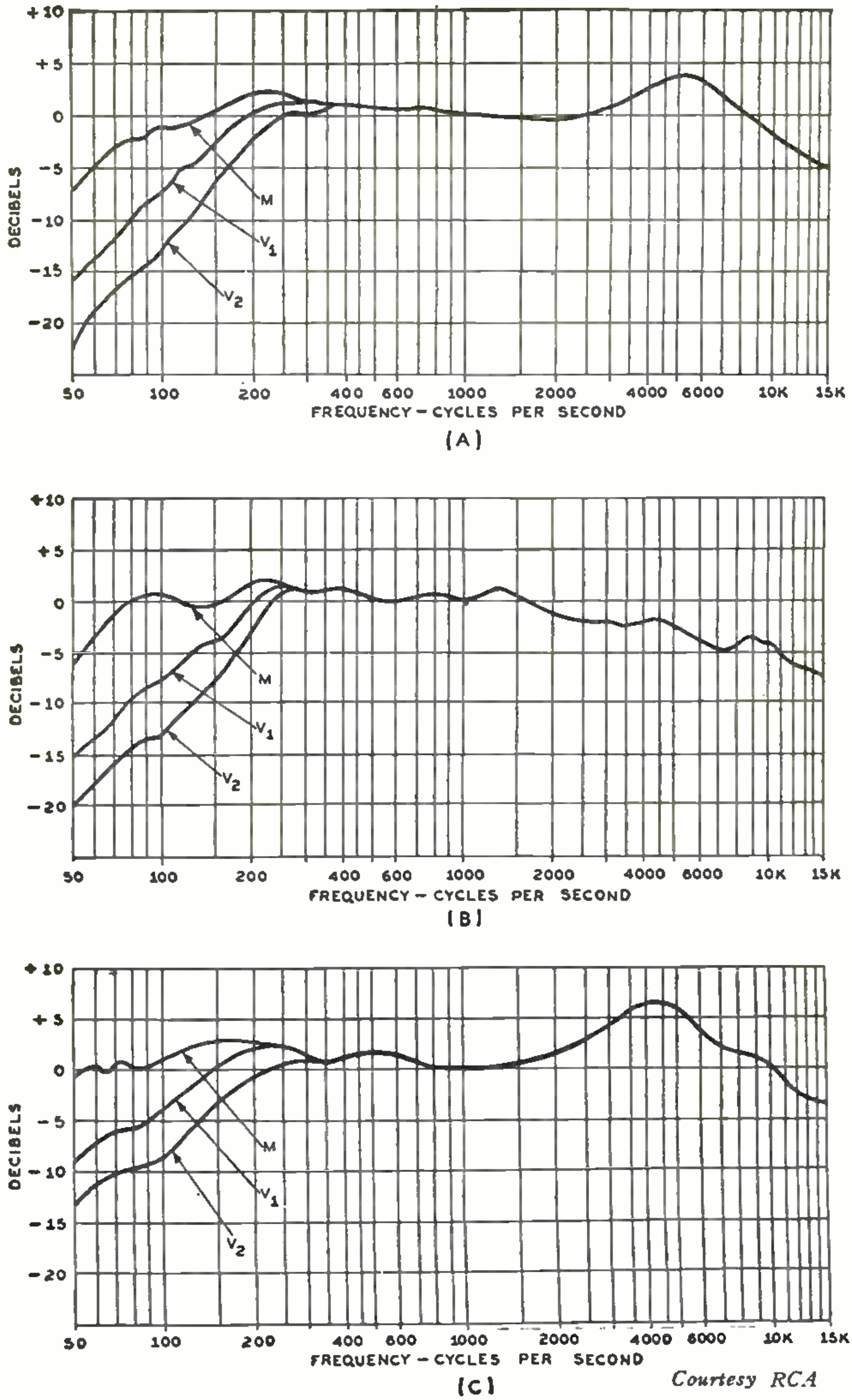
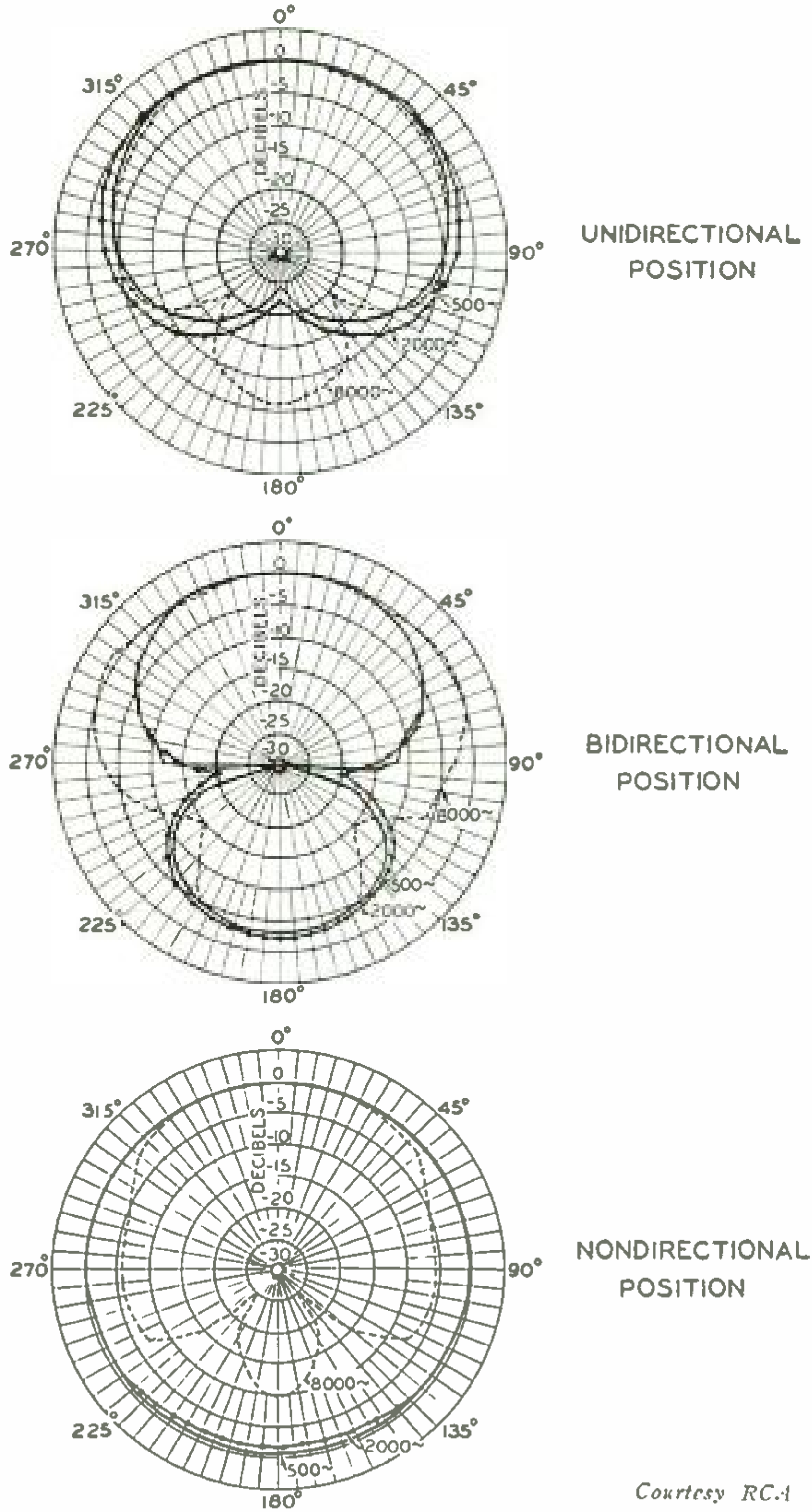


Fig. 18-28. When the switch of the polydirectional microphone is set at "U," the response curve of (A) is obtained. Response curves for bidirectional "B" and nondirectional "N" switch settings are shown in (B) and (C) respectively.



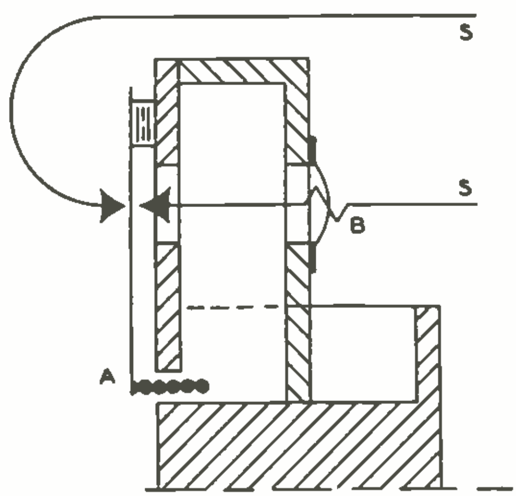
Courtesy RCA

Fig. 18-29. Directional response characteristic curves for the three adjustable shaft positions of the RCA type 77-D microphone. The upper curve corresponds to the "U" position of the shaft; the other curves correspond to the "B" and "N" positions respectively.

The Mechanophase Unidirectional Microphone

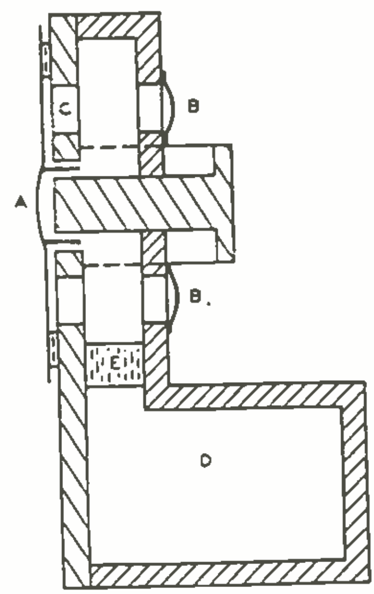
The *mechanophase* principle is one applying only to the Electro-Voice cardioid microphone and is different from any method of obtaining unidirectivity thus far described.

The reader who has followed the text implicitly to this point will understand briefly that the pressure or nondirectional unit has an infinite mechanical impedance looking into the rear of the diaphragm, whereas that of the pressure-gradient or bidirectional, unit is zero. He also understands that any means of combining these two characteristics in the proper manner will annul one loop of the bidirectional pattern resulting in the unidirectional response pattern. This is accomplished in the Electro-Voice mechanophase in a unique way.



Courtesy Electro-Voice, Inc.

Fig. 18-30. Sound *S* transmitted through phase-shifting diaphragm *B* is slowed so that it strikes backside of diagram *A* at same instant that sound *S* from rear traveling around case reaches front of *A*. Because these pressures are equal and opposite there is no movement of the diaphragm for sound reaching it from the rear. Sound striking *A* from the front is reproduced because the sound waves hitting the front and rear diaphragms are not equal and opposite.



Courtesy Electro-Voice, Inc.

Fig. 18-31. Cross-sectional view of the Electro-Voice mechanophase unidirectional microphone, working on the principles described in the caption for Fig. 18-30.

Fig. 18-30 illustrates the basic mechanophase principle of operation. It will be noted from observation of the drawing and the explanatory caption that this unit consists of a dynamic element (associated with diaphragm *A*) and a secondary diaphragm so designed of proper mechanical impedance that it serves as a phasing element to retard the waves from the rear of diaphragm *A* such that this sound energy is cancelled from the microphone output.

Fig. 18-30 is an illustration of one-half of the construction of the total unit as revealed in Fig. 18-31. It is observed here that A is a complete moving-coil unit and the phase-shifting diaphragms B influence each of the active diaphragms, C is the opening between diaphragms, D is the low-frequency resonance chamber, and D is the damping material. This kind of operation gives wide angle frontal response and zero response at rear as typified by the unidirectional pattern.

Fig. 18-32. External view of the mechanophase unidirectional microphone.

Courtesy Electro-Voice, Inc.

Fig. 18-32 shows the appearance of the Electro-Voice Model 731 mechanophase microphone.

SPECIAL-PURPOSE MICROPHONES

Just as the field of radio and sound is a highly complex and varied one, so the application of the sound pickup device ranges through a wide variety of circumstances; some of which require a very specialized operating characteristic. To meet these needs there have been developed a number of special-purpose units so designed as to function in a restricted manner suitable to the special application for which they are intended. Included in this category are highly directional microphones for long-distance work, contact units for electrical amplification of musical instruments, "lapel" microphones to allow freedom of hands and movement of a speaker, "differential" units to cut out background noise, hearing-aid microphones, "reversible" microphones for intercommunication systems, etc. The variety of special-purpose microphones is great, and it behooves the student and technician to become thoroughly acquainted with their characteristics and fields of application.

Parabolic Microphone

It is perhaps only natural that the parabolic reflector was the first principle to be used in attempting to construct a highly directional microphone suitable for such applications as picking up band music from a football field, sounds from a parade at a distance, etc. The reflector idea has been in use for many years in concentrating all types of wave propagation to a point so as to achieve an effective amplification of energy.

It is probably obvious to the reader that a parabola serves to collect the various wavefronts striking the surface, reflecting them to a point of focus much the same as the ordinary magnifying glass concentrates light rays to a point. Fig. 18-33 shows the fundamental principle of a parabolic reflector used with a microphone to achieve a highly directional microphone with reflector concentration of incident sound waves.

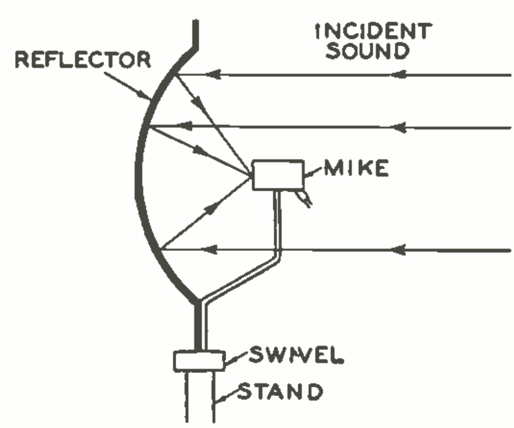


Fig. 18-33. The use of a parabolic reflector to attain unidirectional response. All of the parallel sound waves incident upon the face of the reflector converge at the microphone placed at the focal point of the reflector.

After Olson, Van Nostrand Co.

There are certain inherent difficulties of design in this system that severely limit the practicability of construction. For example, to obtain any appreciable gain in sound pressure at the focal point, the reflector dimensions must be large as compared to the wavelength of the incident sound to be amplified. This is also a necessary requirement if a sharp directional characteristic is to be obtained. While this condition may be achieved easily for the higher audio frequencies of shorter wavelengths, the condition is more troublesome to meet for the lower frequencies of longer wavelengths. To construct a parabolic reflector that would achieve a sharp directional characteristic at frequencies as low as 200 cps would require a size prohibitive to ordinary handling and transportation. It will be found, therefore, that this type of microphone arrangement will have sharper directivity (therefore, higher gain) at the focus for higher frequencies than for lower frequencies. The usual size of a reflector is about 3 feet in diameter.

This extreme high-frequency gain in comparison to lows may be somewhat offset in operation by adjusting the microphone a little displaced from the physical focal point of the parabolic reflector. This will also tend to broaden the high-frequency directional characteristic and achieve a somewhat better over-all frequency response. It should be remembered, however, that the higher frequencies tend to travel in beams, and for some applications this characteristic of the parabolic microphone will be highly beneficial in use. Many listeners to outdoor sports events and football games have noticed the predominance of "boom-boom" in drums of the band and such loss of high frequencies that the music was dull and lifeless. Although music is only incidental to such sporting events, such interludes should maintain the sparkle and excitement of the rest of the broadcast or public-address program.

Line Microphones

The so-called "line microphone" was the succeeding step to the parabolic reflector in an attempt to achieve an efficient directivity for distance work. This principle culminated in the ultradirectional "machine-gun" type to be described in a following paragraph.

As illustrated in Fig. 18-34, the line mike consists of a number of small pipes of varying lengths, terminated at a common junction where a ribbon element is suspended in a magnetic field. This element transforms the acoustical energy received at the open end of the pipes into electrical energy and is itself terminated in an acoustic resistance formed by the familiar damped pipe. This is, then, a pressure-operated transducer in the form of a ribbon actuated by sound energy at the

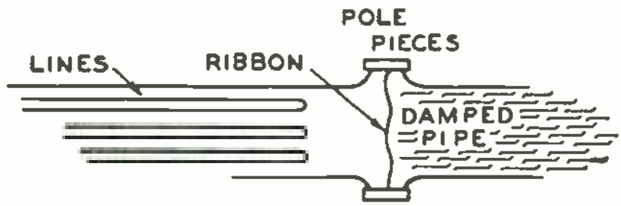


Fig. 18-34. Unidirectional "machine-gun" type microphone. Sound waves coming from the front through the lines or pipes only affect the ribbon.

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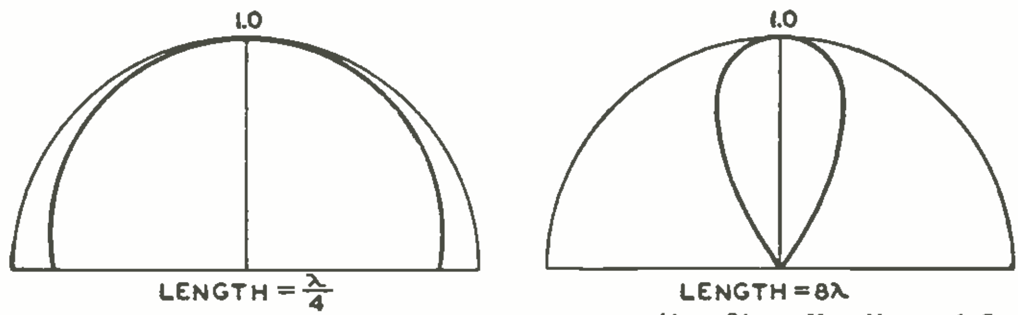

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Fig. 18-35. The directional response curves for two different lengths of line or pipe used in the line microphone to obtain undirectional response. The longer the pipe with respect to wavelength the higher the directivity.

ends of various lengths of pipes. Maximum response is, of course, obtained from the front of the open end of the tubes.

Fig. 18-35 shows the effect of pipe (or line) length on the directional efficiency. As the length of the line becomes greater with respect to wavelength, the directional characteristic becomes greater. This microphone, therefore, in spite of the varying lengths of pipes used, tends to display the old troublesome feature of effect of wavelength on useful directivity, becoming less directional at the lower frequencies.

A means of effectively shortening the necessary length of pipes used for a given directional efficiency is illustrated in Fig. 18-36. It is noted here that the pipes have an inserted acoustic delay. This delay in each pipe is designed of such dimensions that it is proportional to one-fourth times the length of the line. This value varies somewhat in practice between different makes of microphones. Fig. 18-37 shows the directivity of such a line microphone, and illustrates that the directional efficiency is obtained with much shorter length of pipes over that of the simple line mike. Compare with Fig. 18-35. It may be seen, however, that there is some loss of sensitivity in the delay type over the straight-line type, and that the microphone is still influenced by frequency insofar as directional effects are concerned.

Another type of delay-pipe microphone using lines utilizes two sections of sets of pipes such as those of Fig. 18-36, each section terminated on opposite sides of a ribbon element which measures the

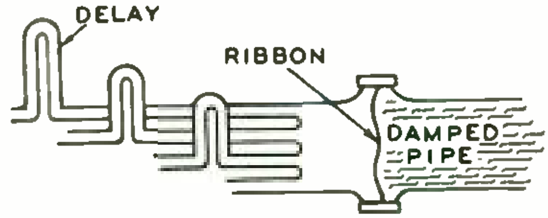
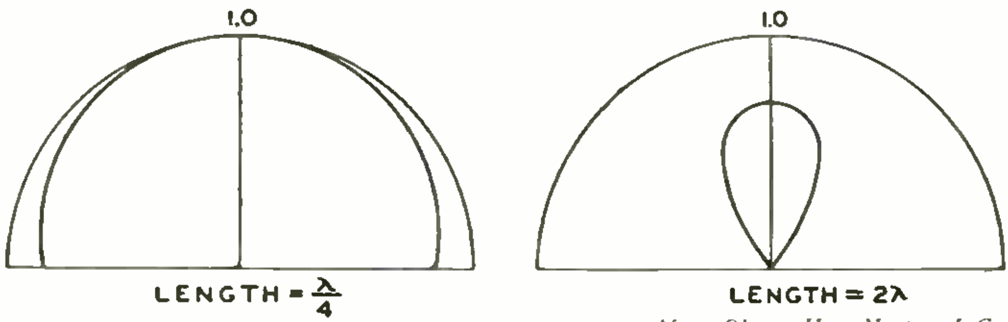


Fig. 18-36. Method of obtaining longer lines for greater low-frequency directivity.

After Olson, Van Nostrand Co.



After Olson, Van Nostrand Co.

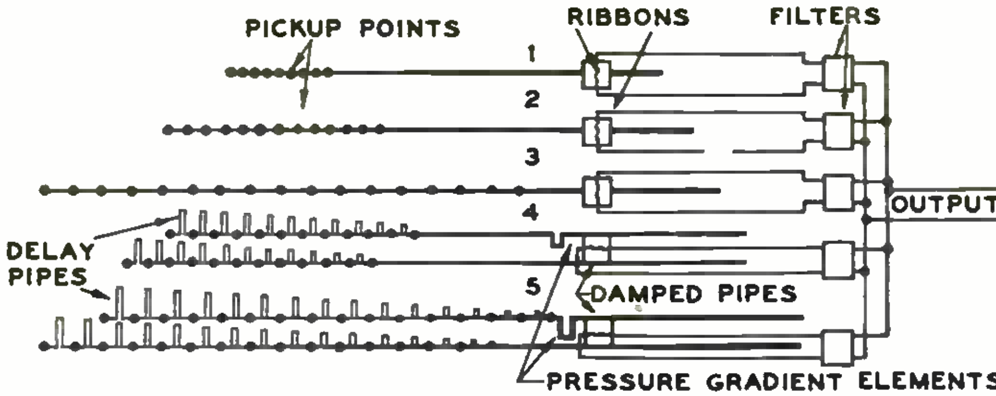
Fig. 18-37. Directional response curves for different pipe lengths using the acoustic delay illustrated in Fig. 18-36. Compare the response for the 2λ line in this figure with the response for the 8λ line in Fig. 18-35.

difference in pressure existing between the two sections. This ribbon serves as a pressure-gradient unit. The directional efficiency of this type is improved even at low frequencies over the two preceding types of line microphone.

Ultradirectional Machine-Gun Microphone

As brought out several times in this text, a microphone to be really useful as a directional instrument should not have a directional efficiency influenced by frequency. Previous discussion in this chapter on the parabolic reflector and the line microphones has shown that the units thus far described, while useful for certain applications, still have the undesirable feature of greater directivity at high frequencies than at the lower frequencies. The microphone to be discussed here is a highly efficient directional microphone at all frequencies between 80 and 8,000 cps. It is a very useful instrument for sporting events, theatrical production, opera, etc., where it is undesirable to run long extensions to the field or to clutter up a stage with numerous microphones necessary to cover a wide area of moving performers with good quality reproduction.

Fig. 18-38 shows the basic method of construction. It consists essentially of lines number 1, 2, and 3 which are of the type shown in Fig. 18-34 and lines 4 and 5 which are of the delay-type lines using a pressure-gradient ribbon described in a previous paragraph. Each



Courtesy Olson, Van Nostrand Co.

Fig. 18-38. Ultradirectional microphone schematic. This microphone uses five lines, three of the straight type and two of the delay type, and both velocity and pressure-gradient microphone ribbons.

line has an associated electrical filter which limits each unit to their respective ranges. Fig. 18-39 shows the very effective directional pattern of this microphone between 80 and 8,000 cps, a frequency range of approximately 6½ octaves.

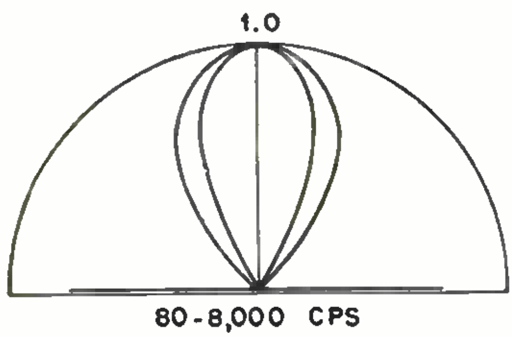


Fig. 18-39. Directional response curves for the ultradirectional microphone shown in Fig. 18-38.

Courtesy Olson, Van Nostrand Co.

Differential Microphones[1]

While all the microphones thus far discussed have had widely varying designs to meet a broad field of application, they have possessed one feature in common; namely, the ability to transform audible sound from a field surrounding the instrument into corresponding electrical impulses. It is obvious, however, that some specialized applications require that only sound extremely close to the microphone be reproduced, in order that a very high background noise level be held far enough below the desired signal that adequate intelligibility of voice waves is obtained. This feature becomes important in many police, aircraft, industrial, marine, and armed forces applications. Broadcasters use this feature for special events from high noise-level points.

[1] Ellithorn, H.E., and Wiggins, A. M., "Antinoise characteristics of differential microphones." *Proc. I.R.E.*, vol. 34, pp. 84P-90P, February, 1946.

The differential microphone was developed to meet these needs. This characteristic dictates the ability to attentuate highly the force actuating the microphone from a distant source, while responding mainly to a sound originating immediately adjacent to the face of the unit.

To be effective, it becomes obvious that a noise level several feet from the mike that results in the same pressure actuating the diaphragm as a speech wave one-quarter inch away, must still be attenuated much more than the speech wave. Thus the differential microphone actually discriminates not against *sound pressure* alone, but against the *distance of origin.* The method by which this is achieved may be understood most clearly by the following explanation.

The differential microphone has two entrances or openings to the sound wave, separated by a distance which may be thought of as an acoustical distance. Fig. 18-40 illustrates the basis of construction for this explanation. Now consider the characteristic of pressure in the air resulting under the stress of a sound wave. The pressure varies inversely as the distance from the source, decreasing rapidly immediately adjacent to the source for a given distance while, for the same unit of distance, decreasing less rapidly at a greater distance. This characteristic as applied to the differential construction is illustrated in Fig. 18-41(A) and 18-41(B).

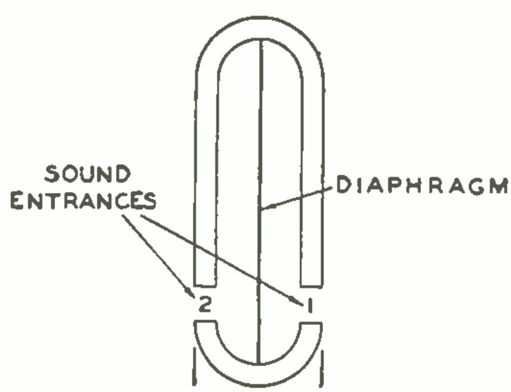
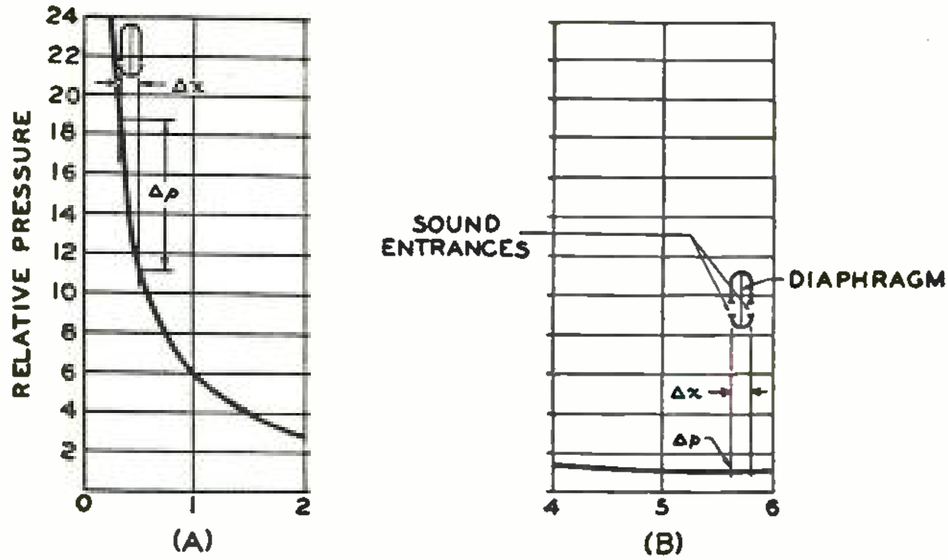


Fig. 18-40. Basis of the differential microphone construction. Such microphones give high speech-to-noise ratios.

Courtesy Electro-Voice, Inc.

It is apparent here that in Fig. 18-41(A) the relative pressure (difference of sound pressure at the two sound entrances) is large in relation to the relative pressure at greater distances as shown in Fig. 18-41(B). In other words, the unit operates on the difference in sound pressure between the two openings, and is a pressure-gradient operated device. In this way, a noise level from a distance behind a speech level that originates immediately adjacent to the unit, even though of the same level at the diaphragm as that of the speech, is



Courtesy Electro-Voice, Inc.

Fig. 18-41. Relative pressure at the two entrances of the differential microphone as a function of the distance of the sound source from the microphone.

reduced 18 to 20 db, resulting in a response of about 100 to 1 of speech in relation to the noise.

The differential microphone, then, is essentially a close-talking device giving good quality and articulation to voice signals even where ambient noise is 100 db or more.

The Choice of Microphone Type

It is because of the extreme variance of conditions imposed upon the wide field of sound transmission that there have been so many types of microphones designed and developed. Thus the choice of a type (or types) of instrument for a particular purpose appears as a very important function in the over-all performance; indeed this factor, in many instances, spells the difference between success and utter failure of a particular project.

Following is a brief outline of the fundamental factors to be carefully considered in choosing a microphone type:

Number of sound sources to be covered
Polar response pattern
Available output level
Output impedance
Length of line necessary
Effect of acoustical conditions encountered
Size and weight of instrument

Ambient noise level at pickup point
Frequency response desired or necessary
Frequency response of mike in relation to associated amplifiers
Type of sound or sounds to be reproduced
Distance necessary to be maintained
Effect of climatic conditions to be met
Cost of microphone in relation to associated equipment.

It is, of course, obvious that many users of a microphone will have to consider the possibility of meeting many different conditions of application with a minimum number of units. It is safe to say that the majority will fall into this classification. Exceptions will be those concerned primarily with only one application, such as an amateur radio installation, mobile or fixed police, fire, marine, or industrial installations, etc.

In any case, the very first consideration is the possible number of sound sources to be covered at one time, and whether this condition will vary from time to time. This condition will determine the necessary polar response pattern to be obtained. The equipment with which the microphone is to be used will determine the necessary available output level, frequency response, output impedance, and logical cost of the microphone. If it is to be used under portable conditions it must be small, light, and more rugged in design. Ambient noise level will determine the necessary polar response pattern and design of acoustical components of the unit. The type of sounds to be reproduced will determine the necessary frequency response. Climatic conditions to be met will fix the type of transducer element.

About as many different conditions as will ever be encountered are included in the field of broadcasting and recording. Recording here is referred to the professional studio, as distinguished from home recording. As a general rule, the recording engineer is concerned only with the studio conditions of microphone choice.

The Studio

As is thoroughly analyzed in Chapter 4, studio acoustics severely affect the effective polar response pattern of a microphone. The number of sound sources to be covered at a given time will vary from one to any number that the studio will accommodate. In general, however, a microphone of wide frequency response and of low output impedance will be necessary. Ambient noise level will be extremely low, and climatic conditions need not be considered here.

It is obvious, then, that a high-quality unit is desirable, and a number of different characteristics in the way of polar response patterns are necessary, ranging from nondirectional to sharply unidirectional. The variable pattern microphones find their most useful application in this field.

In broadcasting "announce" booths used solely for announcements, news, etc., the choice may depend upon a number of operational factors. If, for example, one microphone may be called upon to cover both an announcer and a separate newscaster, the pickup unit must be either nondirectional or bidirectional. Many times a "salt-shaker" mike is used in the nondirectional position for two announcers, and used with a baffle tilted in the semidirectional position for one announcer. The ribbon microphone (bidirectional) may be used just as effectively in most conditions where the booth is not too "live" that might cause a "hollow" effect due to sound reflections from an opposite wall.

In any case, the choice of a microphone for a particular setup will depend upon type of program and effect of acoustics, as considered in detail in Chapter 4 on Operational Technique.

Chapter 19

THE BROADCAST STUDIO

THE ENDEAVOR to realize high-fidelity transmission of broadcast programs is definitely not new; it has been the goal of at least some engineers since the earliest days of broadcasting. The realization of over-all high-fidelity service, however, includes the receiving set in the home, and it has not been until very recently that the "average" set in the medium price market was worthy of the extraordinary efforts of some broadcasters to render high-fidelity service. Conversely, it is apparent at the present time that with a good receiver, noticeable differences in fidelity characteristics of different stations within the range of the receiving position are observed by the critical listener.

If the present state of development in broadcast amplifier equipment is taken as the sole criterion, then high-fidelity transmission is truly here. Frequency response is within 2 db of 1000-cycle reference from 30 to 15,000 cycles, and is limited only by wire-line connecting links in amplitude-modulation (a-m) installations, or not at all in frequency-modulation (f-m) installations. Noise level at the antenna of the transmitter is at least 60 db below 100% modulation, and dynamic range capability is at least 40 db for a.m. and 70 db for f.m. Unfortunately, however, the actual existence of high-fidelity depends on many factors other than the a-f and r-f amplifiers associated with the installation. These amplifiers, according to the ideas of some, form the "heart" of the transmission system insofar as high fidelity is concerned. Actually, they are merely a link in the chain of necessary functions of broadcasting a program, and are no more important to fidelity than the other links, as Fig. 19-1 demonstrates.

In order to focus attention on the possible weak links, by eliminating the amplifiers, as such, there remain: program and talent, production technicians responsible for pickup technique, the studio itself, program producers and announcers, microphones, mixing and switching circuits, control-room, master-control and transmitter operators, wire-lines, feeder systems and matching units, antennas, and the limitations

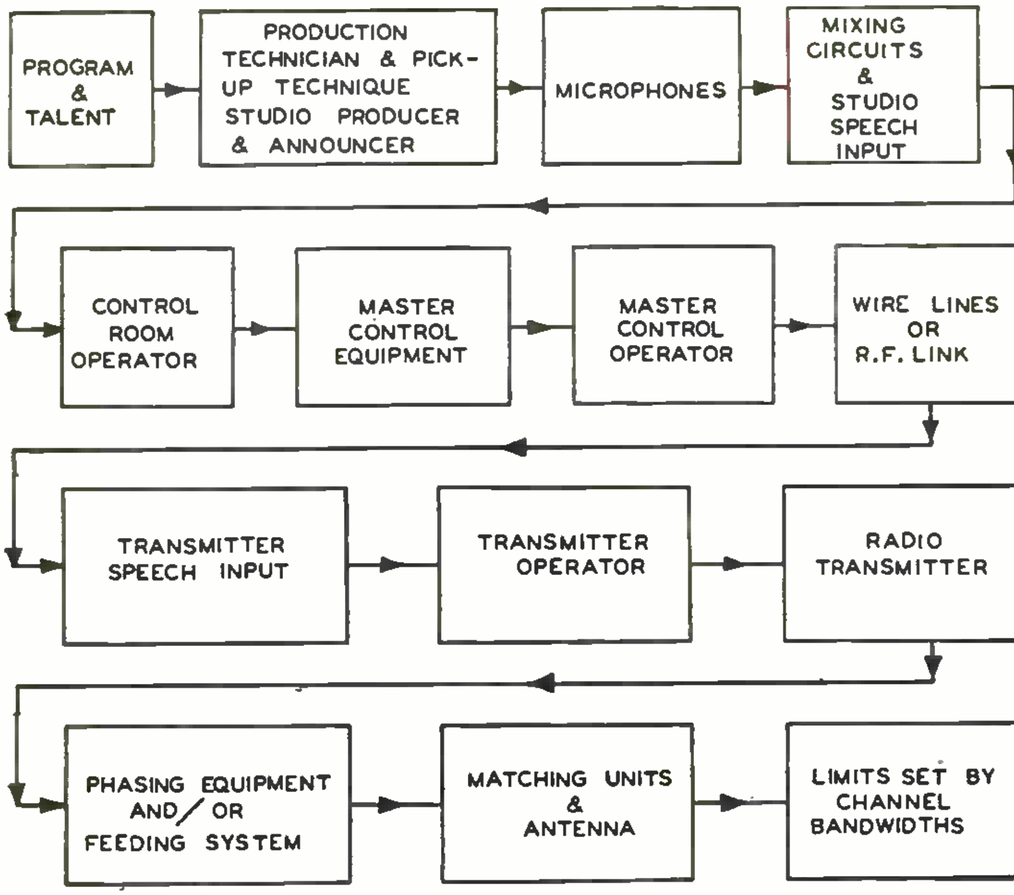


Fig. 19-1. Various links in the chain of putting a program on the air, all of which are important in high-fidelity broadcasting.

set by channel bandwidths subject to government regulations. This presents quite a formidable list, and each item is recognizably inferior to the modern amplifier associated with the broadcast installation in performance. To those familiar with broadcasting, however, it may be shown that the weakest links and those which cause most concern at the present time, are the studio itself, operating personnel, wire-lines, and bandwidth limitations in a-m stations.

The limitations set by wire-line transmission are not serious if considered in relation to the allowable 10-kc channel of the standard broadcast installation. Most lines are equalized to 5000 cycles which is, theoretically, the highest frequency tolerable of any effective strength to prevent adjacent-channel interference. On the other hand, insofar as the relatively small primary coverage area is concerned, the frequency range of modern a-m transmitters (10,000 cycles) if utilized, would allow a marked improvement over present fidelity realization, with class B and C service areas suffering from increased cross talk and interference. Although this situation is a deplorable one, it requires little discussion, in that the problem is primarily one to be solved in the future actions of the FCC.

Thus, there remain two factors to be considered, studio design and operating personnel. It is obvious that the broadcaster could possess high-fidelity equipment from microphone to antenna and still not provide high-fidelity service. In the final analysis, the outcome of any program for a given equipment installation depends entirely on the ability of the technical staff responsible for the operating technique of the equipment. Realizable dynamic range, for instance, which is a highly important factor in high-fidelity transmission, is rarely utilized by station operators. It should be stated here, however, that this is not entirely the fault of operators, but is due rather to a combination of factors including an incomplete correlation between the philosophy of dynamic range and compression amplifiers, inadequate visual monitoring indicators for wide dynamic range, and a confusion of ideas existent among personnel as to the amount of dynamic range tolerable in the home receiver for various types of program content. With the advent of f-m transmission, this problem will become more and more important.

Problems in Studio Design

It is often surprising to discover from a detailed study of the sequence of steps in the development of a certain product, that an indication of a definite direction exists which might well be given the term evolution, and which inevitably indicates a trend that reveals to the searcher an insight into future design of that product. The history of broadcast studio development is interesting not only from this point of view, but also from the viewpoint of establishing the present state of the art as it affects high-fidelity possibilities.

In general the broadcast studio must meet the following requirements:

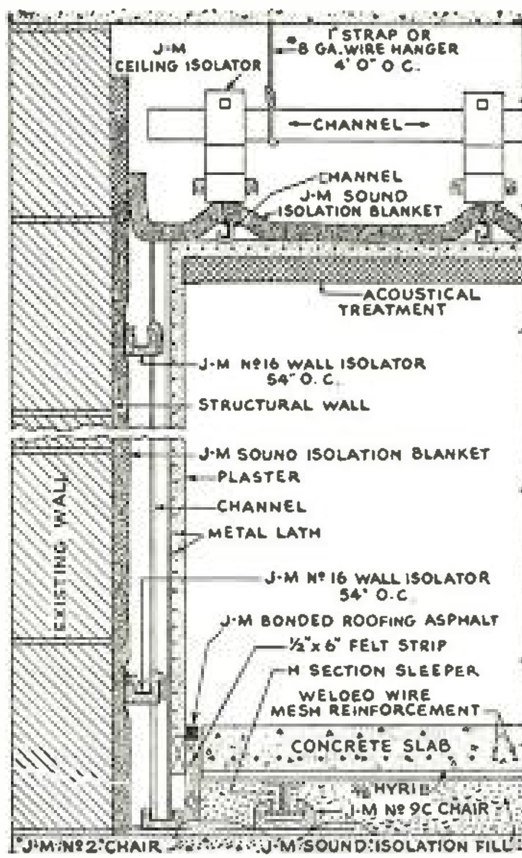
1. Freedom from noise, internal or external
2. Freedom from echoes
3. Diffusion of sound, providing a uniform distribution of sound energy throughout the microphone pickup area
4. Freedom from resonance effects
5. Reverberation reduction such that excessive overlapping of suc-

cessive sound energy of speech articulation or music does not occur

6. Sufficient reverberation such that emphasis of speech and musical overtones is provided to establish a pleasing effect as judged by the listener.

Early Studio Design

In the earliest days of broadcasting, the foremost problems encountered were quite naturally noise and echoes, since "studios" were simply rooms of rectangular shape, with windows of conventional type and walls of ordinary architectural construction. The first steps in



Courtesy Johns-Manville Corp.

Fig. 19-2. Construction details of the "floating studio" for eliminating extraneous noise.

design procedure were then taken to treat the walls acoustically to prevent echoes and "flutter," and to cover the windows with the same acoustical material. This sufficed for a certain era in broadcasting, provided the operator with control over echoes, and practically isolated the microphone from factory whistles, fire sirens, etc. At that time, this type of studio was entirely adequate to satisfy the fidelity requirements of the program transmission possible with associated transmitting and receiving equipment; indeed the electronic amplification of broadcast programs was so much better than the acoustic model phonograph that the general public thought of the radio as a realization of true high-fidelity reproduction.

With the advent of the dynamic loudspeaker, microphone improvements, higher power and wider band amplifiers, the scope of fidelity possibilities began to broaden considerably. Signal-to-noise ratio was improved, and higher volumes could be handled in the receiver without distortion, resulting in a greater dynamic-range capability, but at the same time adding to the burden of studio design, since extraneous noises picked up at the studio were now more noticeable in the home receiver. This fact led to the "floating studio" type of construction, shown in Fig. 19-2.

The period following saw many phenomenal improvements in broadcast equipment in general, such as 100% modulation of the transmitter with greatly reduced distortion, improvement in syllabic transmission characterstics, reduction of spurious frequencies and ripple level, greatly reduced noise levels in switching and mixing circuits, and nonmicrophonic tubes. Yet, strangely enough, studio design remained nearly stagnant over a period of six or seven years except in isolated cases. Indeed, the rectangular shaped, acoustically deadened studio may be recognized by those familiar with the state of the art today as being the most common type of studio among independent stations, even of very recent installation.

The broadcast engineer found himself faced with many apparent difficulties in studios of this type of construction. The big factor, in a room with parallel walls, is the excessive acoustical treatment necessary to overcome the effect of echoes as mentioned previously. This has resulted, in the past, in extreme high-frequency attenuation and a lack of "liveness" such that the brilliancy of musical programs was completely lacking. The loudness intensity for a given meter reading on the volume indicator is very low for a studio of this type in comparison with that obtained from a modern studio.

The effect on speech, while not satisfactory, is not so pronounced as that on music since speech originates within a few feet of the microphone and requires less reverberation to assure naturalness, whereas the space between the source of the music and the microphone is greater, and many things happen to the musical waveforms that must eventually be translated into perceptions of loudness. This effect obviously leads into complex operational difficulties, requiring a lower "peaking" of voice in relation to music on the volume indicator to obtain a comparative loudness intensity in the receiver at home. Furthermore, microphone placement technique for this type studio is such that a number of microphones must be used for a group of performers, since, if a single microphone is employed, a lack of reinforcement of harmonics and overtones of the instruments results in a thin sound, lacking in body.

Another difficulty resulting from parallel-wall construction is shown in Fig. 19-3, where it may be observed that the angle of incidence of

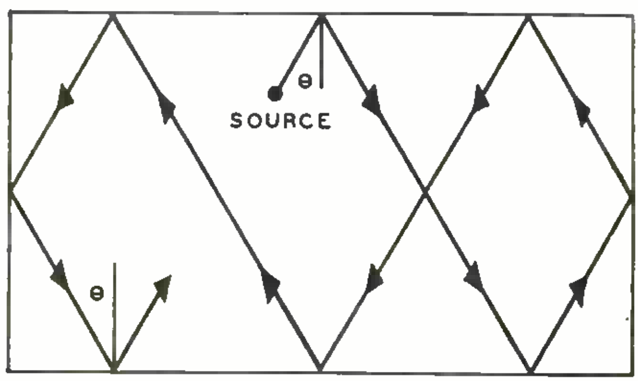


Fig. 19-3. The effect on a sound wave train of parallel-wall construction resulting in undesired resonance effects.
(M. Rettinger: Acoustics in Studios, Proc. IRE, July, 1940. By permission of the Proc. IRE.)

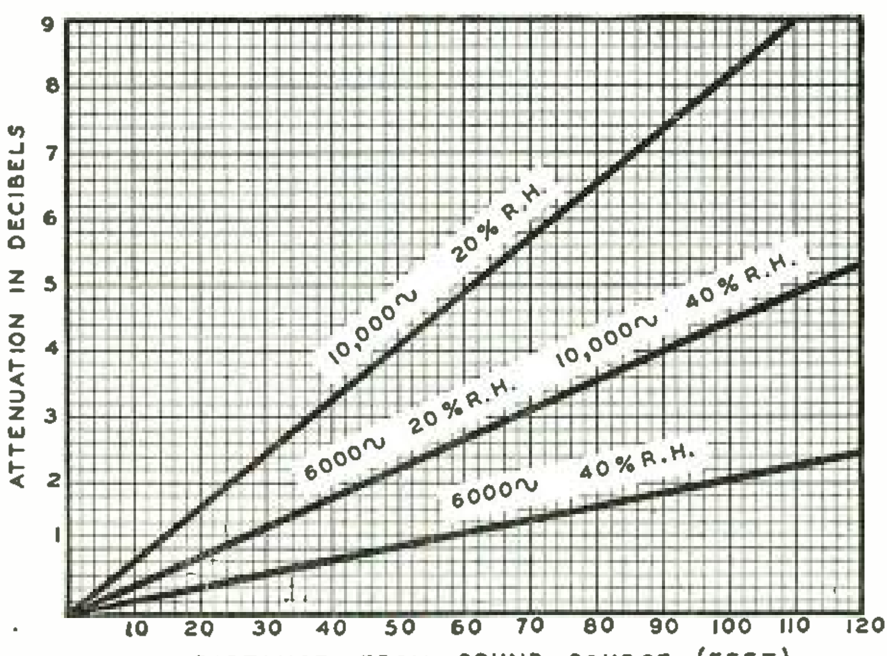
the wave-fronts remains the same no matter how many such reflections occur. Due to the acoustical treatment this reflection (to any great extent) occurs only at the lower frequencies and it may be seen that the nodes would have marked regions of coincident reinforcement, resulting in resonance effects at the lower frequencies, and conditions that would result in diffuse sound distribution are reduced. Thus it becomes obvious that items 3, 4, and 6, as given earlier in requirements for good acoustics, are lacking in studios of this design. In addition, high-frequency response so necessary to brilliancy is reduced, effective dynamic range is inadequate, and operational difficulties are numerous. Thus, it is apparent that the studio becomes the weakest link in the high-fidelity chain in the great majority of broadcast installations today. Exceptions, of course, are the main network studios, and a few independent stations more "production conscious" than the main body of independent broadcasters. It is certainly obvious that the contemplated large scale expansion of f-m service will bring about the need for a revolutionary education in studio requirements for the independent station operator.

Advances in Studio Design

From the foregoing discussion the difficulties to be overcome may be listed as follows:

1. Lack of diffusion of sound
2. Resonance conditions at low frequencies
3. Insufficient reverberation for music
4. High-frequency absorption
5. Critical and multiple microphone placement
6. Operational complexities.

The size and dimensions of the studio comprise a certain problem in studio design since an optimum volume per musician in the studio exists. Reduced to practice, however, this problem becomes one of



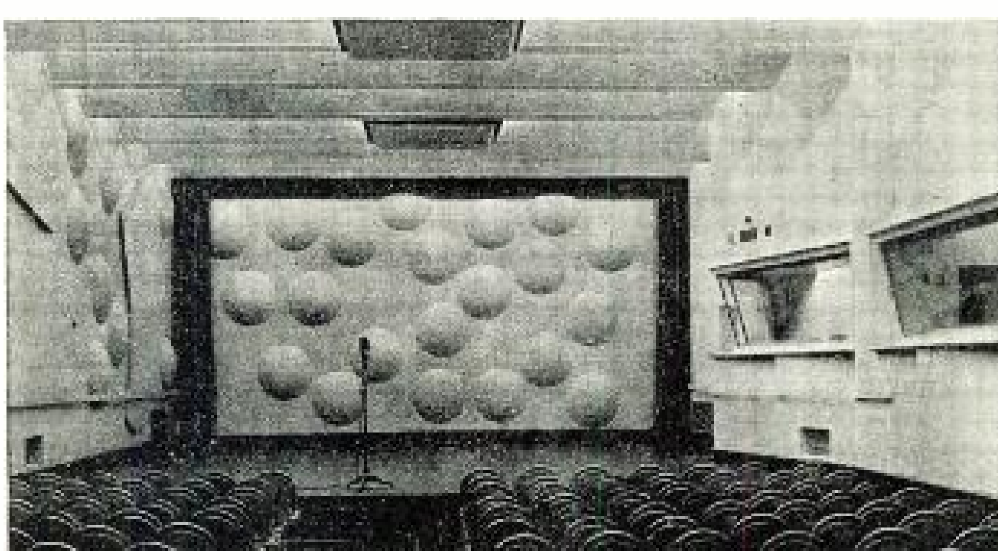
Courtesy Johns Manville Corp.

Fig. 19-4. The absorption by the air of high-frequency sound waves varies with the frequency and the distance from the source.

simply proportioning the studio for a certain maximum number of musicians expected. This is possible because no difficulty exists in obtaining a good pickup of a smaller group in a studio designed for a greater number of musicians; whereas, due to the fact that a small room cannot conveniently be "aurally" enlarged, a large band in a small studio presents a difficult problem. Portable hard "flats" are often used in large studios to enclose a small group of musicians, thus providing the optimum dimensions required for good pickup of a given number of performers.

High-frequency absorption, particularly frequencies of over 5000 cycles, is relatively great as indicated in Fig. 19-4. The absorption of sound by air at these frequencies is actually greater than the surface absorptivity of the studio even under normal temperature and relative humidity conditions. It is not possible to construct a studio having a reverberation time of over 1.2 seconds at 10,000 cycles even with theoretical zero absorptivity in acoustical treatment.[1] This, then, makes obvious the fact brought out before concerning optimum volume per musician in studio design. By distributing the reflector surfaces in proximity to the musical instruments, a maximum of diffused, polyphased high-frequency sound will exist at the microphone without being attenuated injuriously by space in back of the instruments. A minimum number of microphones for adequate pickup is necessary under these conditions.

In general, modern studios are of two types. First is the live-end, dead-end type in which the talent is placed in the live-end and microphones placed in the "microphone area" in the dead-end, thus achieving the correct reverberation of sound waves striking the microphone without bothersome reflections from side and rear walls. This type of studio, as shown in Fig. 19-5, has the advantage of retaining a defi-



Courtesy NBC

Fig. 19-5. An example of a live-end, dead-end studio.

[1] Rettinger, M., "Acoustics in studios," *Proc. I.R.E.*, vol. 28, pp. 296-299, July, 1940.

nite reverberation time not influenced by the size of the studio audience in the dead-end of the studio. It has the disadvantage of limiting the pickup to a definite area in the studio. Second is the general-purpose type studio, consisting of uniformly distributed acoustic treatment, or panels of different type of acoustical elements to achieve a desired condition.

Fig. 19-6 shows graphically the sound-absorbing characteristics of three materials developed by the research department of Johns-Manville in their acoustical laboratory. By proportioning the amount or adjusting the orientation of these three materials in a studio, the time-frequency curve will achieve any desired contour. This type of studio

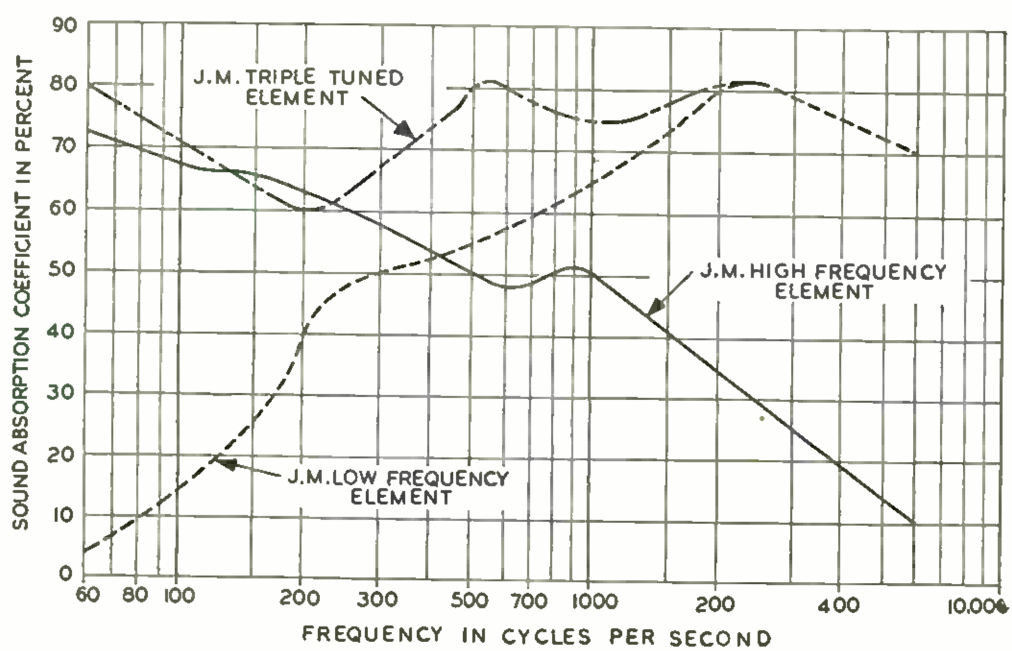


Fig. 19-6, above. The sound absorbing characteristics of three acoustical materials.

Courtesy Johns-Manville Corp.

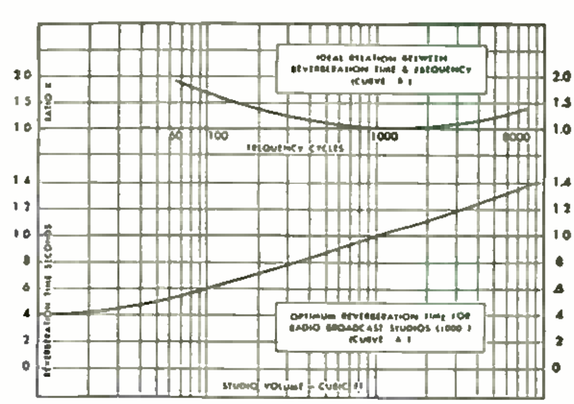


Fig. 19-7. The optimum reverberation time at 1,000 cycles for studios ranging in size from 1,000 to 1.000.000 cubic feet is given in curve "A." Curve "B" gives the relation between reverberation time and frequency.

has the advantage of unlimited pickup area, but has the disadvantage of being affected by size of the studio audience, since a great difference exists in reverberation time when the studio is vacant and when it is occupied by a large group of performers and a large studio audience. Optimum studio reverberation time is shown in the graph of Fig. 19-7.

Studio Finishing Materials

The general-purpose type studio is now generally used in relatively small studios, and announce-transcription booths. There seems to be no advantage to "live-end, dead-end" type of construction for such studios.

The most popular finishing material used in modern studios is a perforated acoustical panel known as "Transite."[2] A panel of this nature is shown in the photo of Fig. 19-8(A). It is a hard material made of asbestos fiber and portland cement. It is backed with a sound absorbent element which may be either studio element, low-frequency

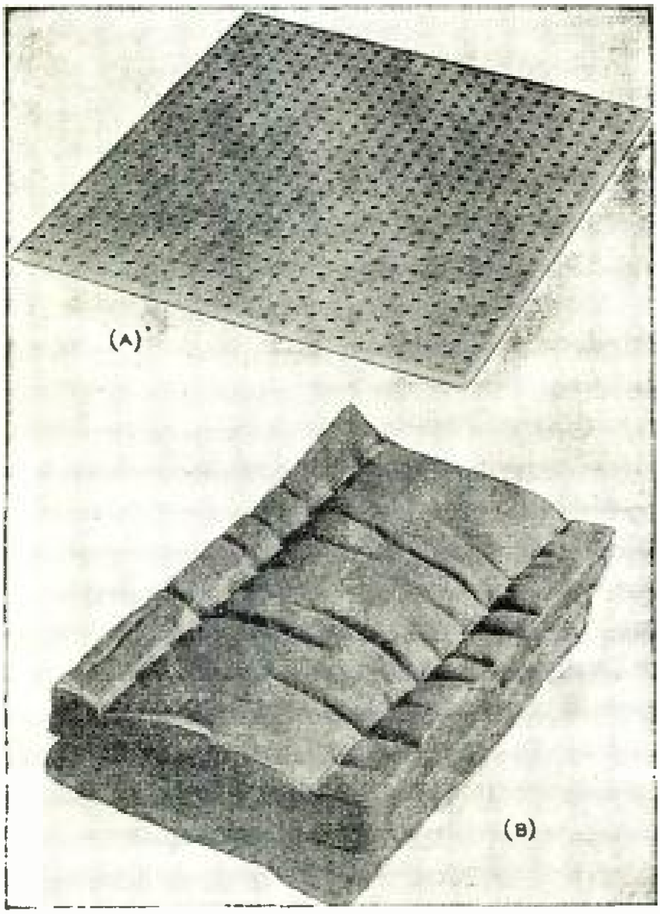
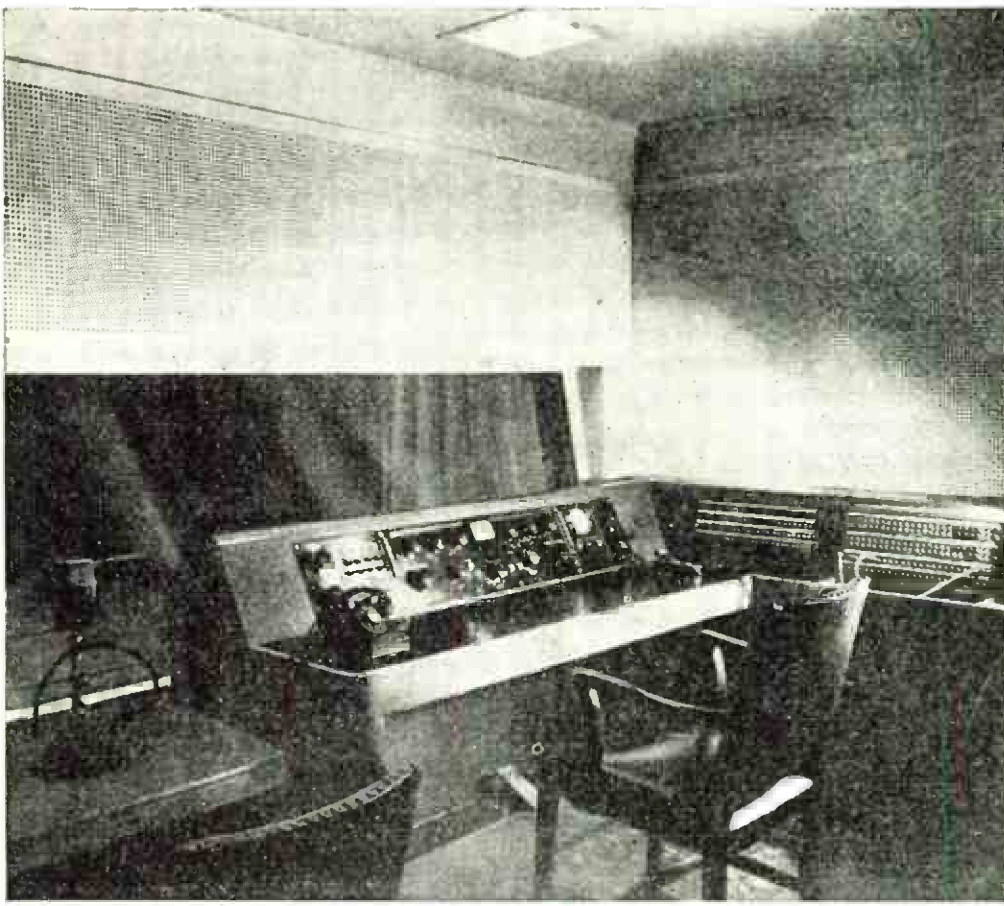


Fig. 19-8. Transite acoustical panel (A) used as the finishing material in modern studios. The rock-wool blanket (B) serves as the sound absorbent backing element.

Courtesy Johns-Manville Corp.

[2] Johns-Manville Corp.

element, rock-wool blankets, etc. A typical rock-wool blanket used with the Transite panel is shown in Fig. 19-8(B). Fig. 19-9(A) illustrates a studio of the general-purpose type at WHAM, with Transite panels on ceiling and on walls down to the lower section. This method is becoming the most popular in modern general-purpose design. Fig. 19-9(B) shows the use of transite in a control room of NBC in Hollywood.

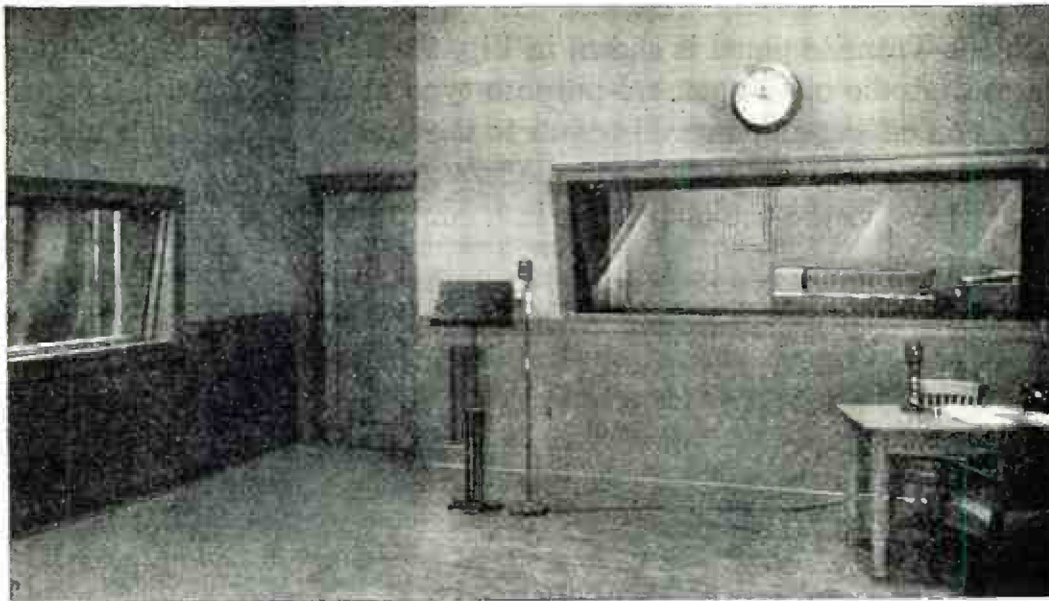


Courtesy Johns-Manville Corp.

Fig. 19-9 (A). A general-purpose type studio at station WHAM with Transite panels on walls and ceiling.

These panels, being perforated, may be cleaned or repainted without altering, to any practical extent, the reverberation characteristics of the studio. Fig. 19-10 shows an auditorium type studio with sawtooth construction finished with Transite, as used at station WHAM.

Fig. 19-11 shows another type of perforated finishing material known as a Sanacoustic unit. This is a perforated metal panel with baked enamel finish and is usually backed with a rock-wool pad. It is often used in transmitter buildings where the reverberation must be controlled for proper monitoring of the program. This type of material

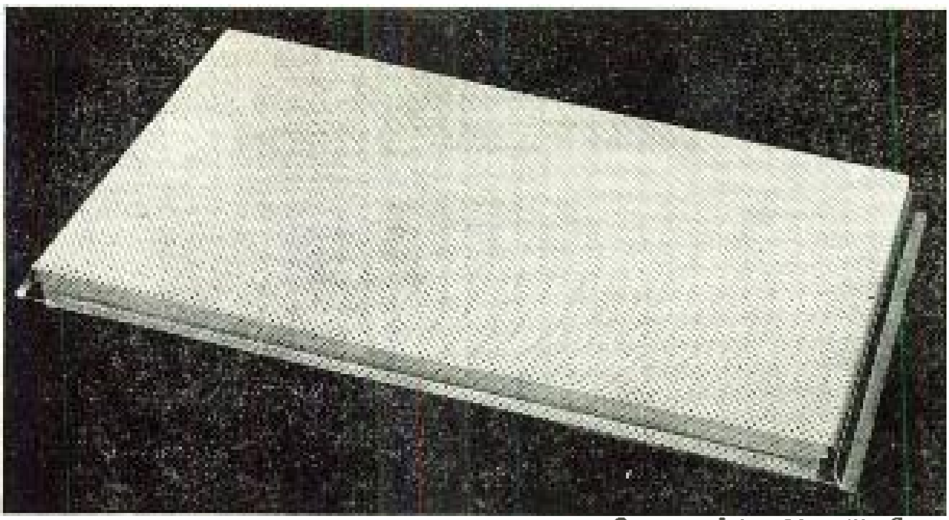


Courtesy Johns-Manville Corp.

Fig. 19-9 (B). NBC in Hollywood also uses Transite in its control rooms (B).

Courtesy Johns-Manville Corp.

Fig. 19-10. This auditorium type studio with sawtooth construction is finished with Transite.



Courtesy Johns-Manville Corp.

Fig. 19-11. A Sanacoustic unit. This acoustical panel is made of perforated metal and backed with a rock-wool pad.

has the advantage of being able to be refinished without altering the reverberation time of the room. Acoustic plaster as formerly used in transmitter rooms cannot be repainted and still maintain sound absorptive characteristics.

Checking Reverberation Time

The original Sabine formula is no longer used for broadcast studios of modern design, being more applicable to theater and public auditoriums. It was found that periods of reverberation time in relatively high sound absorptive rooms such as broadcast studios, using the Sabine formula, was lower in practice than in theory. For reference purposes, this original formula is given here, as it is in a great number of textbooks used

$$T = \frac{0.05V}{a \times S}$$

The new formula used for broadcast studios is known as the Eyring formula, and is stated as follows

$$T = \frac{0.05V}{-\log (1 - a)\, S}$$

where

T = reverberation time
V = volume in cubic feet
a = average coefficient of absorption for interior surface area
S = total area in square feet of interior surface.

When measuring reverberation time at the higher frequencies, consideration must be given to the absorption in the air, as shown in Fig. 19-4. The Morris-Nixon curves of Fig. 19-7 are generally taken as optimum of designers of modern broadcast studios.

Chapter 20

BROADCAST ANTENNA SYSTEMS

CHOICE of the broadcast transmitter site is a highly specialized field that usually comes under the supervision of a consulting engineering firm. A brief outline of the factors affecting the proper location, however, is of prime importance to the serious broadcast employee who likes to have a comprehensive picture of operation and engineering.

In the discussion to follow, it is necessary to keep in mind that field-strength of a radio wave is expressed in "millivolts" or "microvolts per meter." This is a measurement of the stress produced in the ether by the carrier wave that is equivalent to the voltage induced in a conductor one meter in length due to the magnetic flux of the wave sweeping across the conductor at the velocity of light. This field strength is greatly affected by the conductivity of the soil over which it travels. The soil conductivity is expressed in "electromagnetic units," abbreviated emu. This value of soil conductivity varies over a considerable range with the type of soil concerned. Values will be around 3×10^{-13} emu for most loam (good conductivity) and about 1×10^{-14} emu for dry, sandy, or rocky ground which has relatively poor conductivity.

Service Area

The "primary coverage area" of a broadcast transmitter is that area around the towers which provides a distortionless and interference-free signal. This is provided by the ground wave, which must have a carrer-to-noise ratio of at least 18 db, and a field strength of at least several times the strength of the sky wave at the point measured.

The so-called "secondary coverage area" is that area outside the primary area which is supplied primarily by the sky wave. The sky wave, of course, is subject to selective fading (due to changing heights of the Heaviside layer) with resulting distortion effects. The sky wave at broadcast frequencies is almost completely absorbed in the daytime, thus a secondary service area of any appreciable extent appears only at night. Fig. 20-1 shows how the attenuation of the sky wave varies through the sunset period.

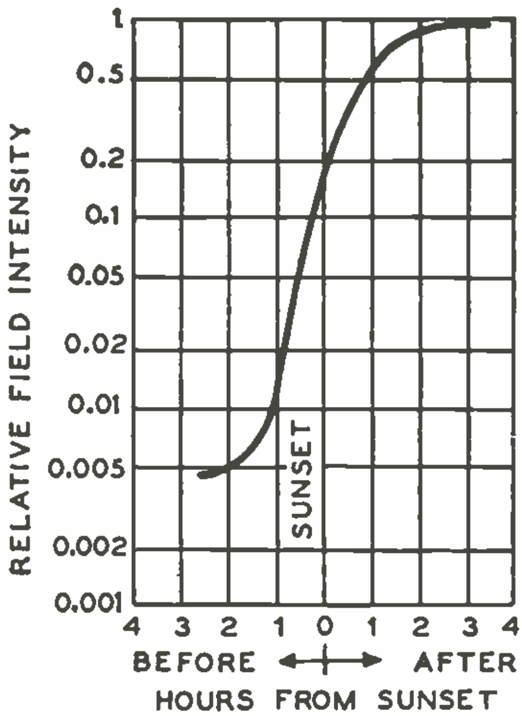


Fig. 20-1. The relative field intensity increases sharply after sunset. Measurements taken at 800 kc over 560 miles during spring.

After FCC

Required Field Strength

Since the required field strength for satisfactory coverage depends on the existing interference level, the value will vary with location. It may be assumed in general that the interference level is greatest in the industrial and business sections of metropolitan areas, less in the residential areas, and still less in rural areas.

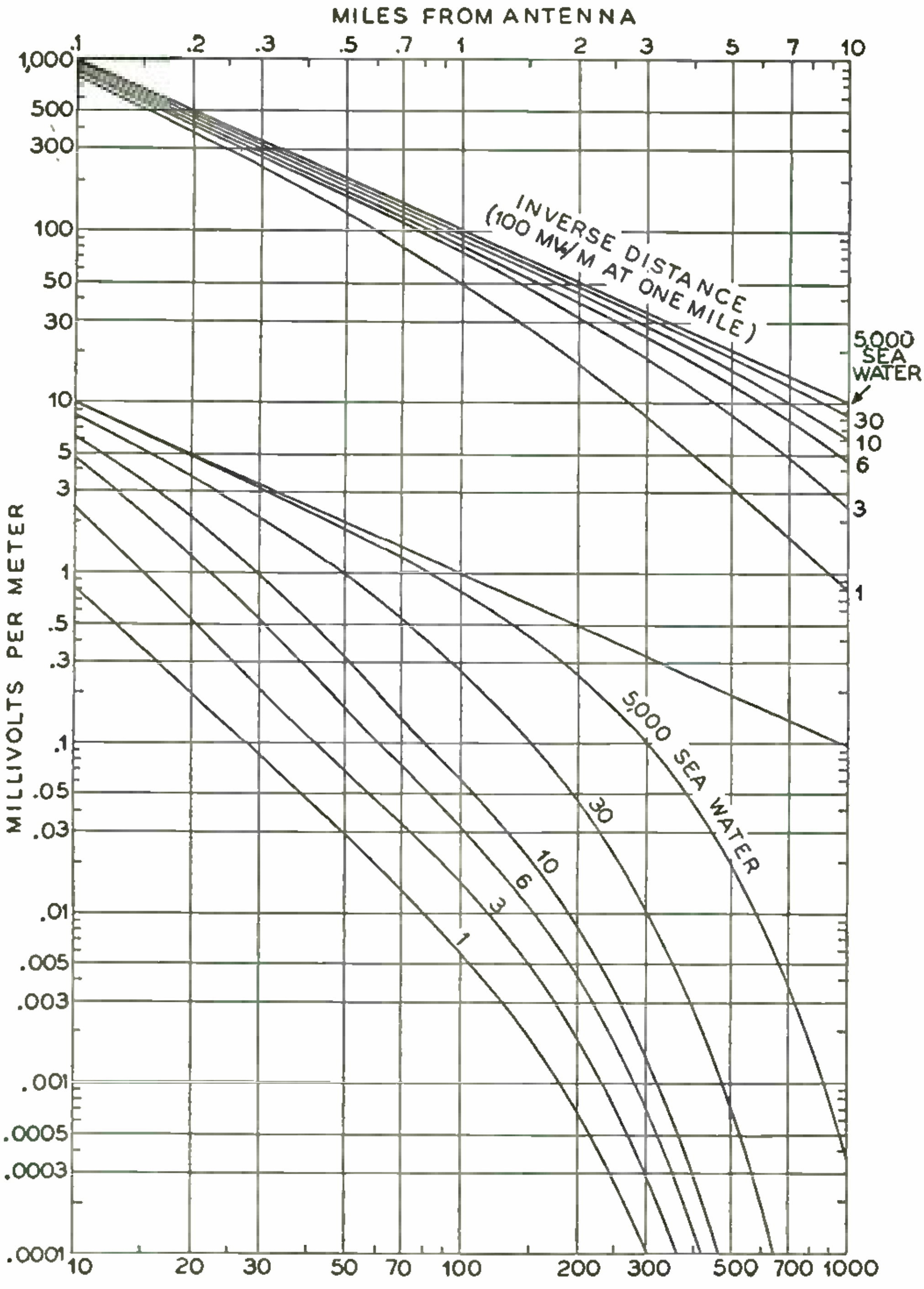
TABLE 1

PRIMARY SERVICE

Area:	Field Intensity Ground-Wave
City business or factory areas	10 to 50 mv/m
City residential areas	2 to 10 mv/m
Rural—all areas during winter or northern areas during summer	0.1 to 0.5 mv/m
Rural—southern areas during summer	0.25 to 1.0 mv/m

—From FCC Standards

Table 1 shows the approxmate field strengths necessary to render adequate service (for primary area) under various conditions. In some locations where conditions are more favorable than average, primary service may be obtained with somewhat weaker field strength than those indicated, and, of course, coverage of an intermittent na-



After FCC

Fig. 20-2. The conductivity of the soil has a great effect on the attenuation of the broadcast signal.

Table 2

PROTECTED SERVICE CONTOURS AND PERMISSIBLE INTERFERENCE SIGNALS FOR BROADCAST STATIONS

Class of Station	Class of Channel Used	Permissible Power	Signal Intensity Contour of Area Protected From Objectionable Interference[1]		Permissible Interfering Signal on Same Channel[2]	
			Day[3]	Night	Day[3]	Night[4]
Ia	Clear	50 kw	SC 100μv/m AC 500μv/m	Not duplicated	5 μv/m	Not duplicated
Ib	Clear	10 kw to 50 kw	SC 100μv/m AC 500μv/m	500 μv/m (50% sky wave)	5 μv/m	25 μv/m
II	Clear	0.25 kw to 50 kw	500 μv/m	2,500 μv/m[5] (ground wave)	25 μv/m	125 μv/m[5]
III-A ..	Regional	1 kw to 5 kw	500 μv/m	2,500 μv/m (ground wave)	25 μv/m	125 μv/m
III-B ..	Regional	0.5 to 1 kw night and 5 kw day	500 μv/m	4,000 μv/m (ground wave)	25 μv/m	200 μv/m
IV	Local[6]	0.1 kw to 0.25 kw	500 μv/m	400 μv/m (ground wave)	25 μv/m	200 μv/m

[1] When it is shown that primary service is rendered by any of the above classes of stations, beyond the normally protected contour, and when primary service to approximately 90 per cent of the population (population served with adequate signal) of the area between the normally protected contour and the contour to which such station actually serves, is not supplied by any other station or stations. the contour to which protection may be afforded in such cases will be determined from the individual merits of the case under consideration. When a station is already limited by interference from other stations to a contour of higher value than that normally protected for its class, this contour shall be the established standard for such station with respect to interference from all other stations.

[2] For adjacent channels see Table 3.

[3] Ground wave.

[4] Sky-wave field intensity for 10 per cent or more of the time.

[5] These values are with respect to interference from all stations except Class Ib. which stations may cause interference to a field intensity contour of higher value. However, it is recommended that Class II stations be so located that the interference received from Class Ib stations will not exceed these values. If the Class II stations are limited by Class Ib stations to higher values, then such values shall be the established standard with respect to protection from all other stations.

[6] Class IV stations may also be assigned to regional channels according to section 3.29.

SC = Same channel.

AC = Adjacent channel.

ture prevails at times in localities where an hour-to-hour variation of interference intensity occurs.

Approval of a transmitter site by the FCC must entail an application which includes a map showing the 250-, 25-, and 5-mv/m contours and the population residing in the 250-mv/m contour (the so-called "blanket area"). This map also indicates by symbols the character of each area (business, manufacturing, residential, etc.), heights of tallest buildings or other obstructions, density and distribution of population, and location of airports and airways. The field-strength contours which would be produced by a transmitter at any particular location, the population within each contour, and the areas where the signal might be subject to nighttime fading and interference, are the determining factors in choosing the most favorable site. For this reason, propagation data that permit prediction of signal attenuation in all directions from a proposed location are of prime importance to the engineer.

TABLE 3

ADJACENT CHANNEL INTERFERENCE

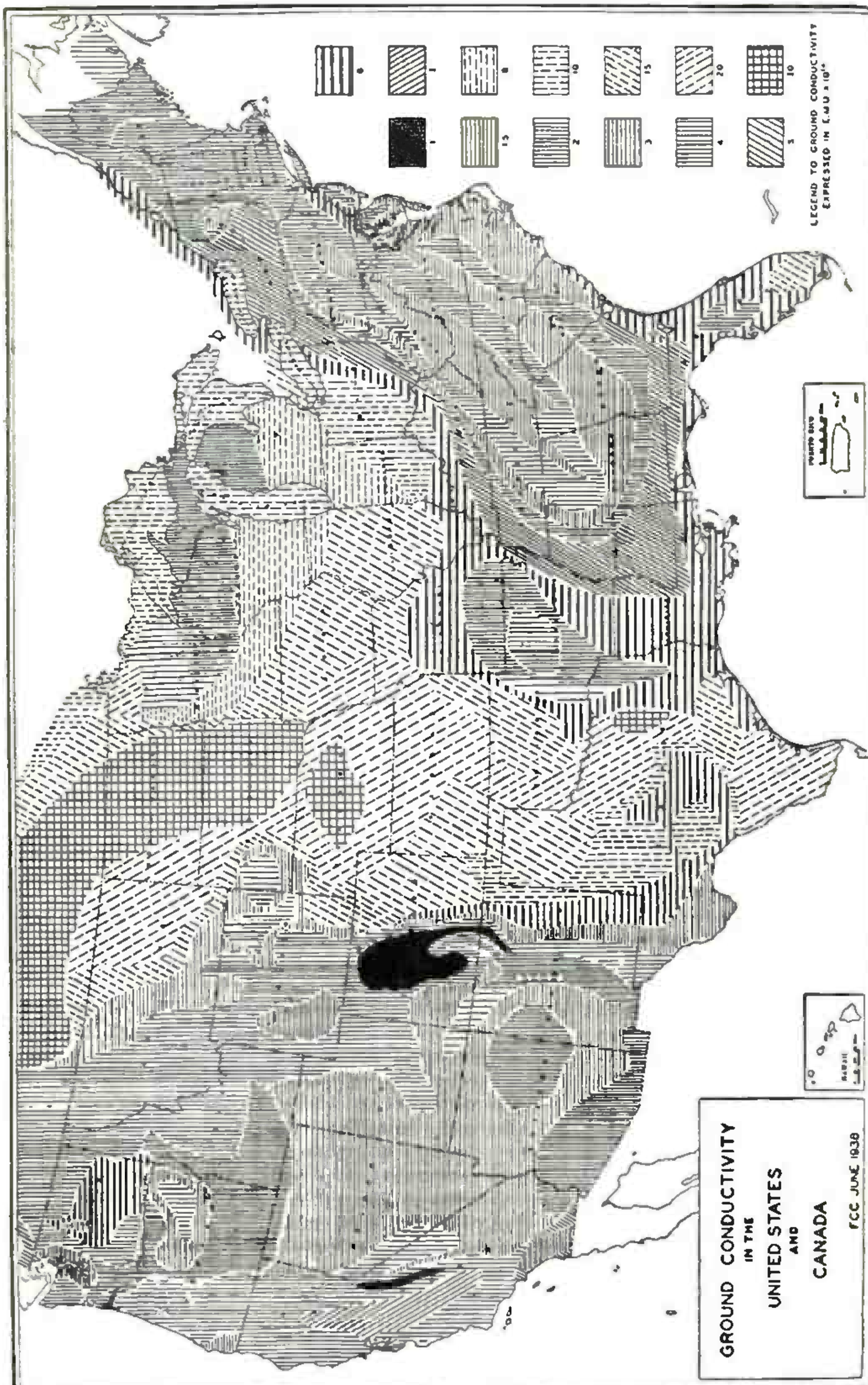
Channel separation between desired and undesired stations:	Maximum Ground Wave Field Intensity of Undesired Station
10 kc	0.25 mv/m
20 kc	5.0 mv/m
30 kc	25.0 mv/m

—From FCC Standards

Ground Wave Propagation Data

The primary service area resulting from a transmitter of given frequency and power depends upon earth conductivity and directivity of the antenna system. The graph of Fig. 20-2 illustrates the effect of soil conductivity on signal attenuation. This type of graph is published by the FCC in blocks of frequencies as shown, some 20 graphs being required to cover the broadcast-band assignments. They show the ground-wave field intensity curve plotted against distance for various conductivity values.

Fig. 20-3 is a map of the approximate and average soil conductivity values for the United States. The protected service contours and permissible interference signals on the same channel for various classes



Courtesy FCC

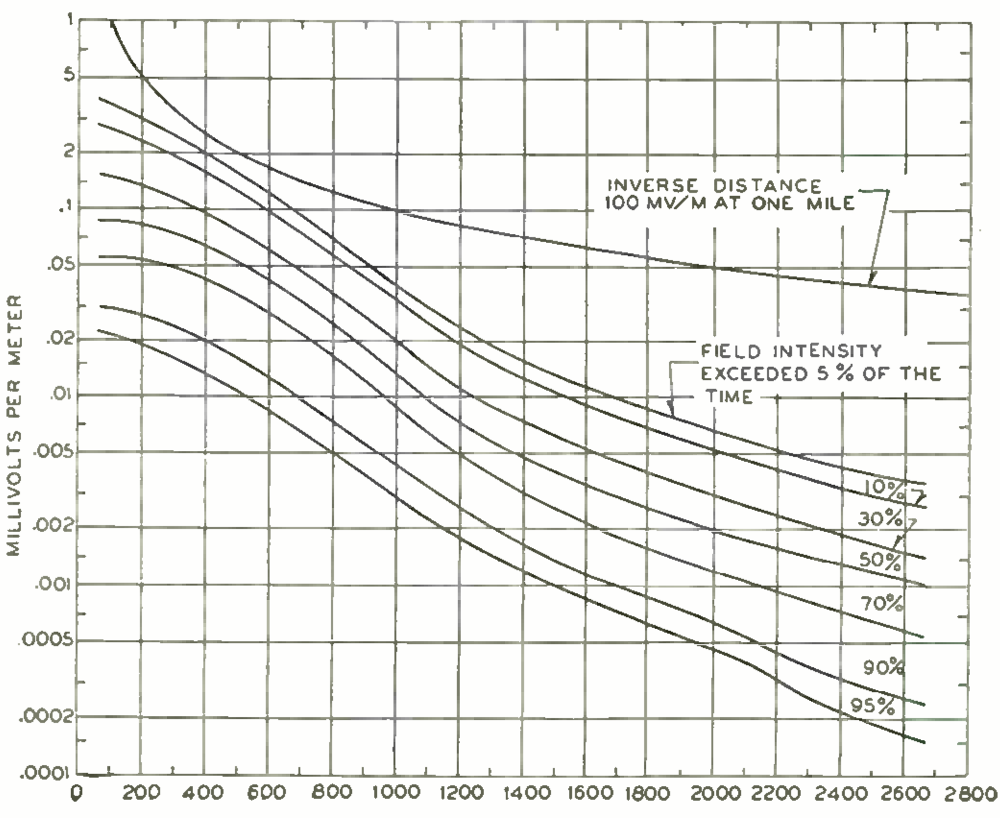
Fig. 20-3. The approximate and average soil conductivity values for the United States.

of broadcast stations are shown in Table 2. Permissible interference levels for adjacent channels is shown in Table 3.

The above curves and tables form the nucleus for gaining necessary information concerning the proposed transmitter site as follows:

Using the Propagation Data

Assume that it is desired to locate a 5-kw class 2 station on 980 kc, 175 miles from a class 2 station of 1-kw power on 990 kc. It is necessary to determine the amount of interference caused by the proposed station to the established 1-kw 990-kc transmitter. Assume also that both stations use nondirectional antennas of such height as to produce an effective field (for 1 kw) of 175 mv/m. Assume further that they are located so that observation of the map of Fig. 20-3 shows an estimated ground conductivity of 6 x 10^{-14} emu. Looking up the required protection to class II during the daytime in Table 2, it may be seen that the protection is to the 500 μv/m contour. The curves of Fig. 20-2 are plotted for 100 mv/m at a mile; therefore, to find the



After FCC

Fig. 20-4. Average sky-wave field intensity, corresponding to the second hour after sunset at the recording station.

distance to the 500 μv/m contour of the 1-kw station, we must determine the distance on the appropriate curve to the proper contour. To determine this proper contour, we must use the following relation:

$$\frac{100 \text{ mv/m}}{175 \text{ mv/m}} \times 500 \ \mu\text{v/m} = 285 \ \mu\text{v/m}.$$

Now. by using the graph of Fig. 20-2 and the curve marked 6 (for 6×10^{-14} emu) it is found that the service area (estimated) of the 1-kw station is about 40 miles. Since $175 - 40 = 135$, we have 135 miles for the interfering signal of the 5-kw station to travel. Again using the appropriate curve of Fig. 20-2, it is found that the signal from the 5-kw station at 35 miles would be 62.5 μv/m. Since the stations are separated by 10 kc, the interfering signal may have a value up to 250 μv/m allowable by the FCC, as shown in Table 3.

The principles, as outlined above, are not used when the sky wave of the interfering station is in excess of 5 times the established signal for 10% or more of the time (with frequency separation of 10 kc). When this condition prevails, the interference must be estimated on the basis of the sky wave also, and the propagation curve of Fig. 20-4 for sky wave signals must be considered.

Other Factors

Other considerations than effective conductivity of the area surrounding the proposed site must be taken into account. Following are excerpts from the FCC Standards of Good Engineering Practice which outline these extra factors:

As a guide, the Engineering Department has established certain engineering principles based on the extensive experience of the Engineering Department and all data available along this line, including those presented at the informal engineering hearings of October 5, 1936, January 18, 1937, and June 6, 1938.

The four primary objectives to be obtained in the selection of a site for a transmitter of a broadcast station are as follows:

1. To serve adequately the center of population in which the studio is located and to give maximum coverage to adjacent areas.
2. To cause and experience minimum interference to and from other stations.

3. To present a minimum hazard to air navigation consistent with objectives 1 and 2.
4. To fulfill certain other requirements given below.

TABLE A

Power of Station	Population of City or Metropolitan Area[1]	Approximate Radius of Blanket Area 250 mv/m[2]	Site—Distance From Center of City (Business or Geographical)	Maximum Percentage of Total Population in Blanket Area[1]
		Miles	*Miles*	*Percent*
100 watts..........	5,000–50,000	0.15	½–1	1
100 watts..........	50,000 or more	0.15	([3])	...
250–500 watts......	5,000–150,000	0.3–0.5	1–3	1
250–500 watts......	150,000 or more	0.3–0.5	([3])	...
1 kilowatt..........	5,000 to 200,000	0.6–0.9	2–5	1
1 kilowatt..........	200,000 or more	0.6–0.9	([3])	...
5–10 kilowatts......	All	1.5–2.5	5–10	1
25–50 kilowatts.....	All	3.0–4.5	10–15	1

[1] The total population is the population of the city sought to be served except in those instances when the station is to be located in an area classified by the Department of Commerce, Bureau of Census, as a metropolitan area, in which case the population of the metropolitan area shall apply: *Provided, however*, That when the power of the station is such that all the metropolitan area cannot be served, the population that will actually be served shall determine. The population figures are those determined by the latest official census and where greater population is claimed, the burden of proof is on the applicant.

[2] These radii are only approximate and the actual blanket area (area within the 250 mv/m contour) may be materially different depending on the antenna employed and other factors.

[3] In these instances it is usually necessary to locate the station within the city in order to render satisfactory service throughout the city. Such sites shall be in or near the center of the business district and under no circumstances will a site in the residential area be approved.

—From FCC Standards

Table A is offered as a general guide to be used in determining the approximate site of broadcast transmitters.

In case the power and the population of the city are such that it should be located at some distance from the center of the city, the approximate distance is given as well as the population of the so-called "blanket area." The "blanket area" of a broadcast station is defined as that area adjacent to the transmitter in which the usual broadcast receiver would be subject to some type of interference to the reception of other stations due to the strong signal from the station. The normal blanket area of a broadcast station is that area lying within the 250-millivolt-per-meter contour line. The average radii of the blanket areas for broadcast stations of the various powers are given in Table A.

In those cases where it is impossible or impractical to locate a station in accordance with the above specifications, the Commission will give consideration to approving locations where not more than 1% of the population (as above specified) is included within the 500-millivolt-per-meter contour, provided the applicant submits an affidavit setting forth the reasons why the normal specifications cannot be complied with, and further that the applicant will assume full responsibility for adjustment of any reasonable complaints arising from the excessively strong signals of the applicant's station. Particular attention must be given to avoiding cross modulation.

In this connection, attention is invited to the fact that it has been found very unsatisfactory to locate broadcast stations so that high signal intensities occur in areas with overhead electric power or telephone distribution systems and sections where the wiring and plumbing are old or improperly installed. These areas are usually found in the older or poorer sections of a city. These conditions give rise to cross-modulation interference due to the nonlinear conductivity characteristics of contacts between wiring, plumbing, or other conductors. This type of interference is independent of the selectivity characteristics of the receiver and normally can be eliminated only by correction of the condition causing the interference. Cross modulation tends to increase with frequency and in some areas it has been found impossible to eliminate all sources of cross modulation, resulting in an unsatisfactory condition for both licensee and listeners.

Broadcast station transmitters will not be permitted to be located in these areas even though the population is within the requirements of Table A, unless the licensee assumes full responsibility for, and it appears it can adjust all complaints satisfactorily.

If the city under consideration is of irregular shape, the station is of high power, a directional antenna system is employed, or if other unusual conditions obtain, the table may not apply and special consideration must be given. However, the general principles set out will still apply.

In selecting a site in the center of a city it is usually necessary to place the radiating system on the top of a building. This building should be large enough to permit the installation of a satisfactory ground and/or counterpoise system. Great care must be taken to avoid selecting a building surrounded by taller buildings or where any near-by building higher than the antenna is located in the direction which it is desired to serve. Such a building will tend to cast "radio shadows," which may materially reduce the coverage of the station in that direction. Irrespective of the height of surrounding buildings, the building where the antenna is located should not have a height of approximately one-quarter wavelength. A study of antenna systems located on buildings tends to indicate that where the building is approximately a quarter wavelength in height, the efficiency of radiation may be materially reduced.

TABLE B

Type of Terrain	Inductivity	Conductivity	Absorption Factor at 50 Miles 1000 kc[1]
Sea water, minimum attenuation	81	4.64×10^{-11}	1.0
Pastoral, low hills, rich soil, typical of Dallas, Tex., Lincoln, Nebr., and Wolf Point, Mont., areas	20	3×10^{-13}	0.50
Pastoral, low hills, rich soil, typical of Ohio and Illinois	14	10^{-13}	0.17
Flat country, marshy, densely wooded, typical of Louisiana near Mississippi River	12	7.5×10^{-14}	0.13
Pastoral, medium hills, and forestation, typical of Maryland, Pennsylvania, New York, exclusive of mountainous territory and sea coasts	13	6×10^{-14}	0.09
Pastoral, medium hills, and forestation, heavy clay soil, typical of central Virginia	13	4×10^{-14}	0.05
Rocky soil, steep hills, typical of New England	14	2×10^{-14}	0.025
Sandy, dry, flat, typical of coastal country	10	2×10^{-14}	0.024
City, industrial areas, average attenuation	5	10^{-14}	0.011
City, industrial areas, maximum attenuation	3	10^{-15}	0.003

[1] This figure is stated for comparison purposes in order to indicate at a glance which values of conductivity and inductivity represent the higher absorption. This figure is the ratio between field intensity obtained with the soil constants given and with no absorption.

—From FCC Standards

If from Table A it is determined that a site should be selected removed from the city, there are several general conditions to be followed in determining the exact site. The table gives the approximate distance from the center of the city. Three maps should be given consideration if available:

1. Map of the density of population and number of people by sections in the area.
2. Geographical contour map with contour intervals of 20 to 50 feet.
3. Map showing the type, nature, and depth of the soil in the area with special reference to the condition of the moisture throughout the year. (See Table B.)

From these maps a site should be selected that is approximately the required distance from the city, with a minimum population in the "blanket area," and with a minimum number of intervening hills between it and the center of the city. In general, because of ground conditions, it is better to select a site in a low area rather than on top of a hill, and the only condition under which a site on top of a hill should be selected is that only by this means is it possible to avoid a substantial number of hills between the site and the center of a city with the resulting radio shadows. If a site is to be selected to serve a city which is on a general sloping area, it is generally better to select a site below the city than above the city.

If a compromise must be made between probable radio shadows from intervening hills and locating the transmitter on top of a hill, it is generally better to compromise in favor of the low area, where an efficient radiating system may be installed which will more than compensate for losses due to shadows being caused by the hills if not too numerous or too high. Several transmitters have been located on top of hills, but as far as data have been supplied, not a single installation has given superior efficiency of propagation and coverage.

The ideal location of a broadcast transmitter is in a low area of marshy or "crawfishy" soil or area, which is damp the maximum percentage of time and from which a clear view over the entire center of population may be had and the shadow of the tall buildings in the business section of the city would be cast across the minimum residential area.

The type and condition of the soil or earth immediately around a site is very important. Important, to an equal extent, is the soil or earth between the site and the principal area to be served. Sandy soil is considered the worst type, with glacial deposits and mineral-ore areas next. Alluvial, marshy areas, and salt-water bogs have been found to have the least absorption of the signal. One is fortunate to have available such an area and, if not available, the next best condition must be selected.

Table B indicates the values of inductivity and conductivity which it is recommended be used for various types of country in the absence of surveys over the particular area involved. Naturally, values obtained from the use of these figures will be only approximate and should, if possible, be replaced by actual measurements in the area under consideration.

Careful consideration must be given to selecting a site so that the number of people in the blanket area is a minimum. The last column of Table A gives the percentage of the total population of the city or metropolitan area that may be permitted in the blanket area. In general, broadcast transmitters operating with approximately the same power can be grouped in the same approximate area and thereby reduce the interference between them.

If the city is of irregular shape, it is often possible to take advantage of this in selecting a suitable location that will give a maximum coverage and at the same time maintain a minimum of people within the blanket area. The maps giving the density of population will be a key to this. The map giving the elevation by contours will be a key to the obstructing hills between the site and city. The map of the soil conditions will assist in determining the efficiency of the radiating system that may be erected and the absorption of the signal encountered in the surrounding area.

Another factor to be considered is the relation of the site to airports and airways. There are no regulations or laws with respect to distance from airports and airways, but a distance of 3 miles from each is used as a guide. In case a suitable location is found at less distance than this, it may be satisfactory if the towers are suitably painted and lighted in conformity with the requirements of the Civil Aeronautics Administration, or if the towers are not higher than the surrounding objects. The latter is normally considered poor engineering practice; however, in selecting a site the local aeronautical authorities should always be consulted if there is any question concerning erecting a hazard to aviation, and in case of towers over 200 feet high this should always be done. In passing on a location and antenna installation, the Engineering Department refers each case to the Civil Aeronautics Administration for its recommendation. The action of the Administration will be materially expedited by the district airline inspector and local representatives of the airports and airlines forwarding their approval or comments to the Civil Aeronautics Administration, Washington, D. C.

In finally selecting the site, consideration must be given to the required space for erecting an efficient radiating system, including the ground or counterpoise. It is the general practice to use direct grounds consisting of a radial buried-wire system. If the area is such that it is not possible to get such a ground system in soil that remains moist throughout the year, it probably will be found better to erect a counterpoise. (Such a site should be selected only as a last resort.) It, like the antenna itself, must of course be designed properly for the operating frequency and other local conditions.

While an experienced engineer can sometimes select a satisfactory site for a 100-watt station by inspection, it is necessary for engineers of a higher power station to make a field-intensity survey to determine that the site selected will be entirely satisfactory. There are several facts that cannot be determined by inspection that make a survey very desirable for all locations removed from the city. Often two or more sites may be selected that appear to be of equal promise. It is only by means of field-intensity surveys taken with a transmitter at the different sites or from measurements on the signal of near-by stations traversing the terrain involved, that the most desirable site can be determined. There are many factors regarding site efficiency that cannot be determined by any other method.

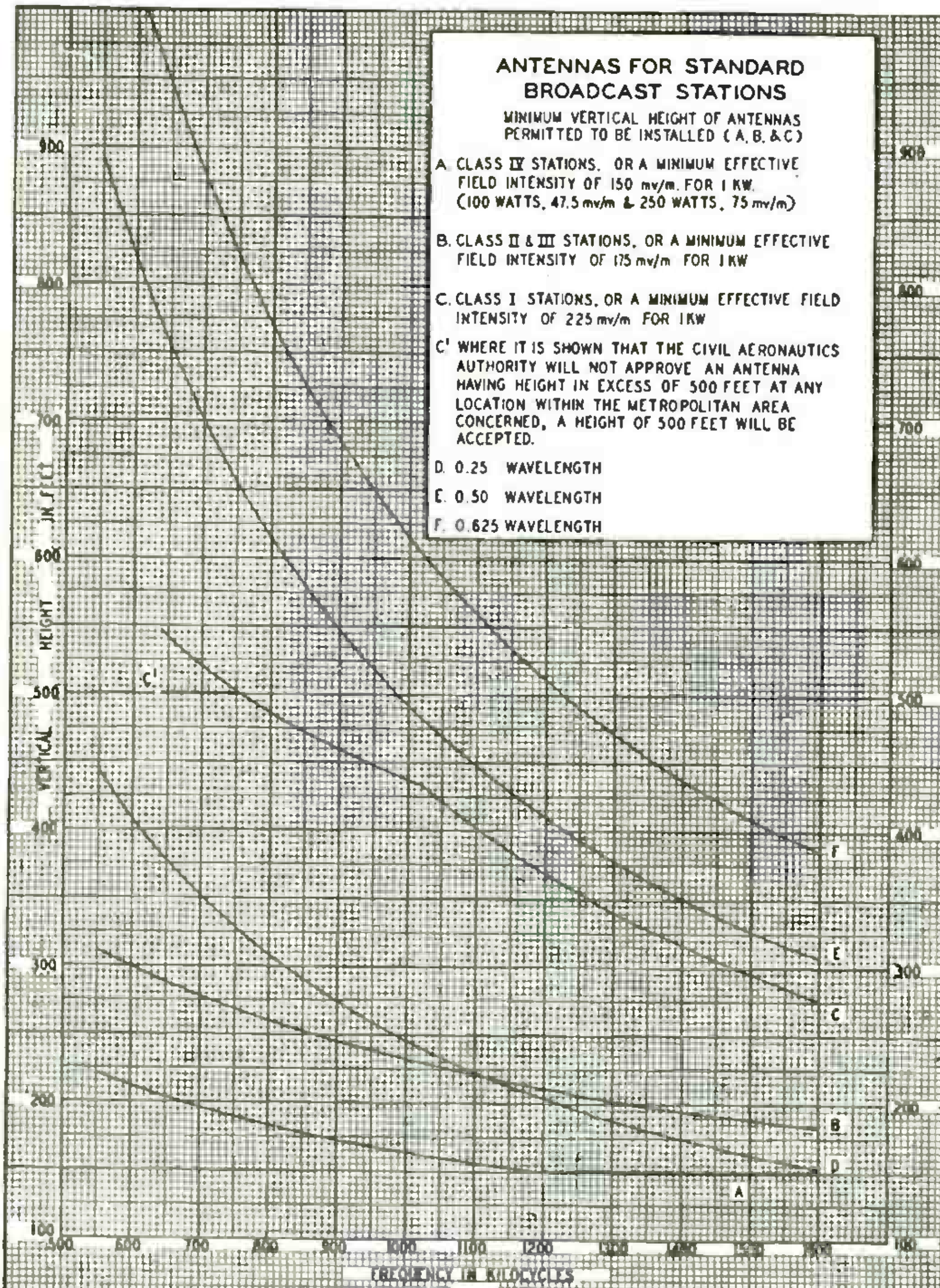
The site selected should meet the following conditions:

1. A minimum field intensity of 25 to 50 millivolts per meter will be obtained over the business or factory areas of the city.
2. A minimum field intensity of 5 to 10 millivolts per meter will be obtained over the most distant residential section.
3. The absorption of the signal is the minimum for any obtainable sites in the area. As a guide in this respect the absorption of the signals from other stations in that area should be followed, as well as the results of tests on other sites.
4. The population within the blanket radius (250 mv/m) does not exceed that specified by Table A.

When making the final selection of a site, the need for a field-intensity survey to establish the exact conditions cannot be stressed too strongly. The selection of a proper site for a broadcast station is an important engineering problem and can only be done properly by experienced radio engineers.

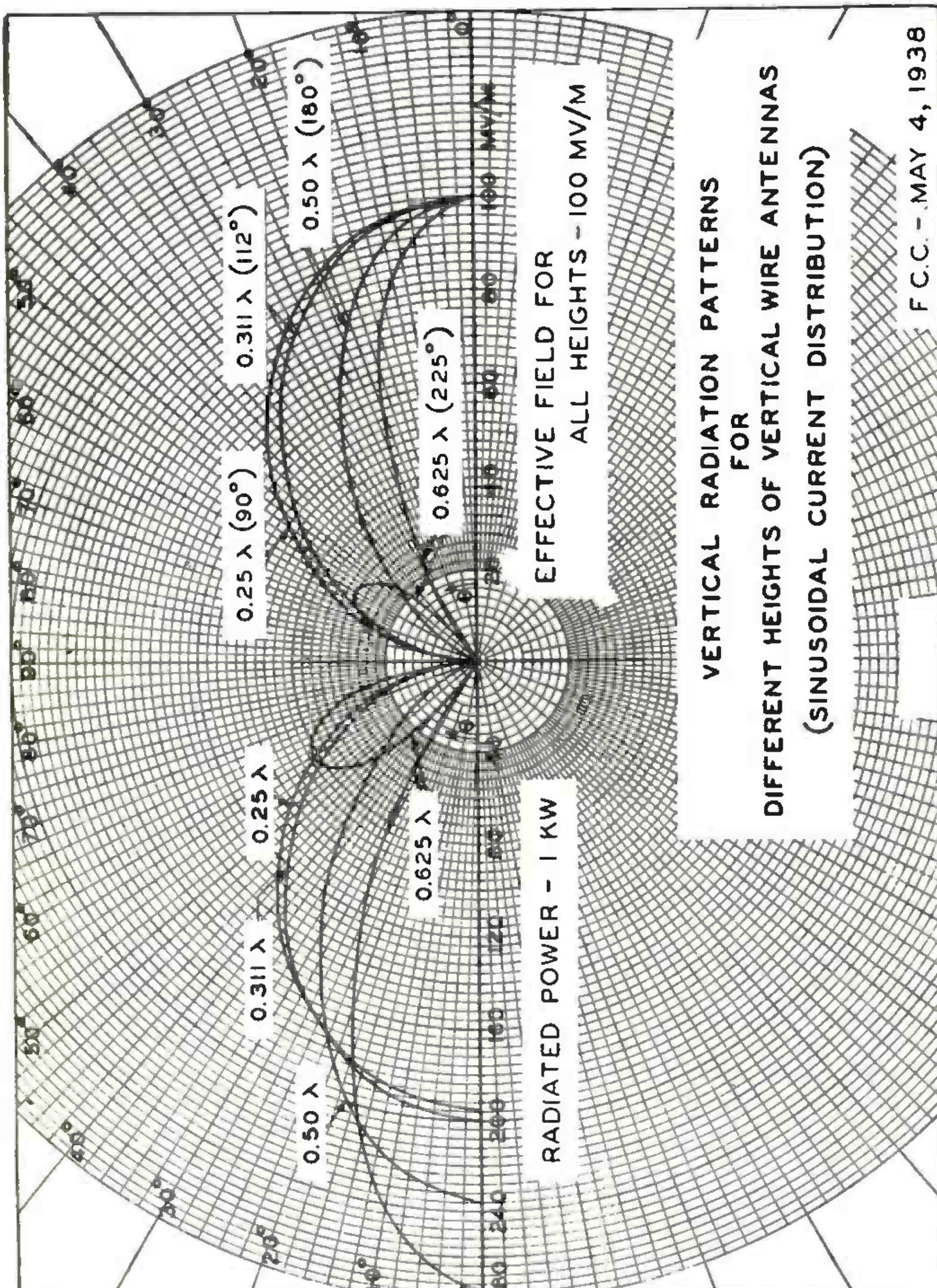
BROADCAST ANTENNA SYSTEMS

As was pointed out in the previous discussion of transmitter location problems, the efficiency of service depends principally upon four



Courtesy FCC

Fig. 20-5. Curves A, B, and C show minimum antenna height needed to deliver the required field intensity for different class stations. Curves D, E, and F show antenna heights for fractions of wavelengths between 500 to 1600 kc.



Courtesy FCC

Fig. 20-6. Comparative vertical radiation patterns for antennas of 0.25, 0.311, 0.5, and 0.625 wavelengths.

factors: frequency of operaton, operating power, ground conductivity, and orientation of transmitter with respect to distribution of population. A fifth factor, namely the design of the radiating system, will also affect the over-all efficiency, especially in areas at a distance from the particular transmitter site.

When application is made to the FCC for new, additional, or modified broadcast facilities (such as changing transmitter location), the applicant must specify the nature of the radiating system to be employed. This system must comply with efficiency standards adopted by the FCC to meet the requirements of good engineering practice for the particular class of service concerned.

To this end, the FCC has set up standards which specify a minimum effective field intensity for any particular class of station. Fig. 20-5, curves A, B, and C show the minimum actual physical height of antennas deemed necessary to deliver the required field intensity for the class station involved.

Observation of these curves show the following requirements:

Curve A. Class IV stations, a minimum height of 150 feet (for frequencies 1200 kc and higher) or a minimum effective field intensity of 150 mv/m for 1 kilowatt (100 watts 47.5 mv/m and 250 watts, 75 mv/m).

Curve B. Class II and III stations. Minimum effective field intensity of 175 mv/m for 1 kilowatt.

Curve C. Class I stations. Minimum effective field intensity of 225 mv/m for 1 kilowatt.

Curves D, E, and F show the physical heights of the antenna for 0.25, 0.5, and 0.625 wavelengths for any frequency from 500 to 1600 kc.

Considerations in Antenna System Design

Some interesting points are involved in the design applications of radiating systems for broadcast frequencies. Fig. 20-6 illustrates the comparative vertical radiation patterns for antennas of 0.25, 0.311, 0.5, and 0.625 wavelengths. Observation of this figure reveals that although an antenna of 0.625 wavelength has a large low-angle lobe, a secondary lobe exists at a higher angle which decreases the effective fade-free area. Fading occurs when the sky wave caused by the reflected energy meets the ground wave and tends to cancel out the signal due to phase reversal.

It has been found in practice, for example, that the strength of the ground wave at a given distance is increased only a few decibels by increasing the height of the antenna from 0.125 to 0.5 wavelength, but the *effective* fade-free area is greatly increased due to the reduction in strength of the high-angle radiation producing a sky wave that returns to ground close to the transmitting location. Increased directivity in the horizontal plane is the main purpose of using higher physical wavelengths over a quarter wavelength antenna. It has been found that an antenna of 190° or 0.53 wavelength is the most efficient height to use where cost of such installation is warranted by conditions involved.

An adequate ground system must be employed with the broadcast antenna in order to obtain maximum efficiency. The FCC specifies that where the vertical radiator is used with the base on the ground, a ground system must be employed consisting of buried radial wires at least one-quarter wavelength long. They require at least 90 such radials, and recommend 120 radials of 0.35 to 0.4 wavelength spaced 3°. In case of high base voltage (such as occurs in antennas approaching 0.5 wavelength) a base screen of adequate dimensions should be employed to prevent high dielectric losses.

REMOTE ANTENNA-CURRENT INDICATOR[1]

1. DESCRIPTION

The Remote Antenna-Current Indicator is designed to give relative indications of the currents in antenna arrays employing up to three elements. The unit provides a means for insuring the correct current relationship between elements, and hence, proper field patterns. It is furnished with a standard rack-mounting panel.

Each of the three current-measuring circuits are wired as shown in the schematic diagram of Fig. 20-7. The 79-ohm plug-in resistor provides the correct termination for sampling lines having surge impedances between approximately 72 and 82 ohms. Plug-in terminating resistors suitable for use with sampling lines having other surge impedances can be obtained on separate order if desired. In general, an exact line match is not necessary, and except for critical line lengths, such as multiples of one-quarter wave, variations of as much as ten per cent from the exact value should introduce no appreciable error in either current or phase indications. The plug-in resistors are located on the rear of the unit.

[1] Courtesy RCA.

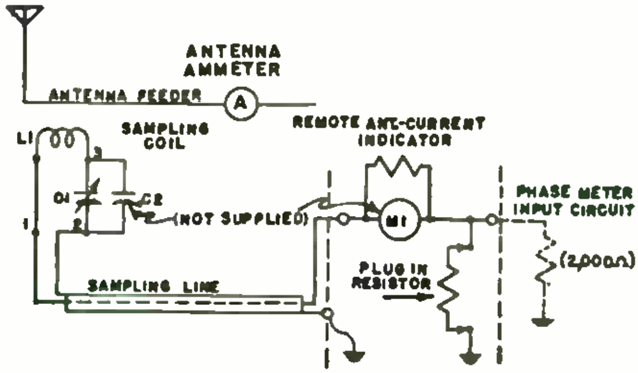


Fig. 20-7. Schematic diagram of the remote antenna-circuit indicator.

2. INSTALLATION

The equipment is designed for mounting in a standard 19-inch rack. If it is to be used in conjunction with the Phase Monitor, the Remote Antenna-Current Indicator should be mounted either directly above or below the monitor in order that short, direct open bus connections can be made between the two instruments. When this is not feasible, and distances up to 3 or 4 feet are involved, interconnections should be made with concentric line. The capacitance per foot of the interconnecting cables should not exceed 30 $\mu\mu$f. Interconnecting lines of equal length should be used in the sampling line circuits for best accuracy in making phase measurements. Leads should be dressed away from each other to avoid cross-coupling.

The three line terminals at the left rear of the unit provide for connection of three sampling lines. The output terminals at the right provide for connecting the unit to three respective terminals (A_2, B_2, and C_2) of the Phase Monitor. If the Phase Monitor is not used, the output terminals can be left open. Three resistors are included to simulate the load imposed by the input circuit of the WM-30A. When the Phase Monitor is used, these resistors should be removed.

3. SAMPLING ADJUSTMENT

The relative advantages and disadvantages of various methods of sampling, and the construction of suitable sampling lines and coils are described later. Reference should be made to this, particularly if the sampling system to be installed is intended to feed a phase monitor.

No means is provided in the equipment for adjusting the panel-meter indication. The desired deflection is obtained experimentally by varying the coupling between the sampling coil and the antenna.

Between 5 and 10 volts output is normally required at the output of the sampling coil to provide sufficient excitation for the Phase Monitor and the Remote Antenna-Current Indicator. The coil output voltage can be estimated with reasonable accuracy when the antenna current is known

$$\text{Output volts, } E = fNIrK$$

where

f = frequency in megacycles
N = number of turns
I = antenna current in amperes
r = radius of coil in inches
K = constant derived for distance (d) and radius (r) as given in Fig. 20-8.

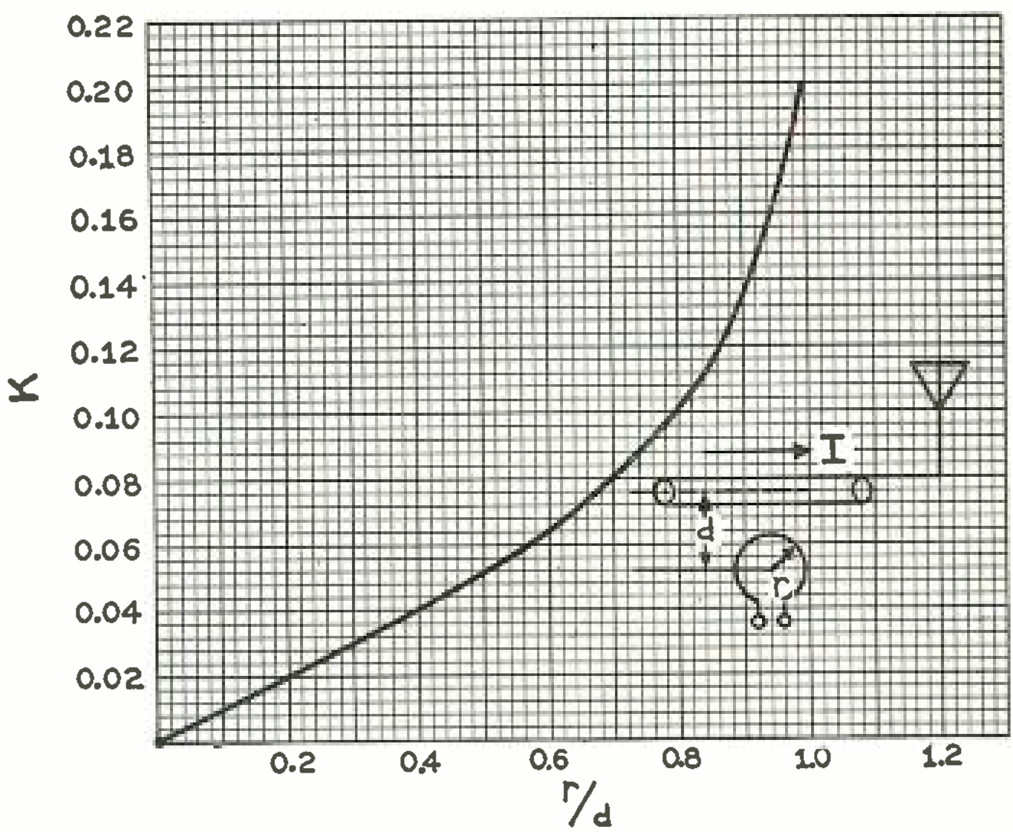


Fig. 20-8. Curve for determining the constant K in the formula for calculating the output voltage.

For example, at a frequency of 1,000 kc and with an antenna current of 4 amperes,

if $d = 4$ inches
$r = 1.5$ inches
$N = 54$ turns

then, $E = (1)(54)(4)(1.5)(0.04) = 13.0$ volts (approx.)

Slight rotation of the coil from maximum coupling will, in this case, give the desired output voltage.

METHODS OF SAMPLING

This section is a detailed treatment of various methods of sampling, together with their relative advantages and disadvantages as applied to different antenna installations.

1. USE OF TUNED CIRCUITS

a. General

The tuned sampling coil can be coupled at virtually any point along the radiator or feed line in many installations and still pick up enough voltage to operate the current and phase indicators. Considerations involved in locating the coil are discussed in Paragraph 3. The coil can be made small, and if a phase meter is not used, the pickup voltage can be conveniently varied over a limited range by slightly detuning the circuit, rather than by varying the degree of coupling which, in some installations, presents a mechanical problem. The tuned coil has one disadvantage for use with a phase meter. Unless the coil is kept precisely tuned to the transmitter frequency, errors will result in phase measurements due to the reactive component in the sampling unit.

b. Location

Since the tuned circuit is more sensitive than a nonresonant loop, care must be exercised in its location. Otherwise, misleading indications may be produced by pickup from adjacent towers or voltages induced by other inductors in the installation. Pickup from an extraneous field will cause error in phase indication, and it may also produce nonlinearity in current indication; therefore, current indications will be in error when a change of operating conditions occurs.

c. Use of the RCA Sampling Kit MI-8217

In general, electrostatic shielding is essential to assure stability and accurate indications. In the RCA Sampling Kit (MI-8217), no provision need be made for electrostatic shielding since each sampling coil is constructed with an internal double electrostatic shield. If the sampling equipment is to be located within a tuning house, a shield compartment as outlined in Fig. 20-9 can be constructed of copper or copper-lined steel. For outdoor locations, the enclosure can be weatherproofed by the use of Type 306 5⅜″ ceramic bowl insulators for powers up to 5 kw and for radiators whose operating impedances are less than approximately 200 ohms at the sampling point. For higher power, larger insulators should be employed. For 50-kw installations, where the radiator is sampled at a point of high impedance, the clearance of the sampling coil from the antenna bus should be increased an additional inch to prevent voltage breakdown.

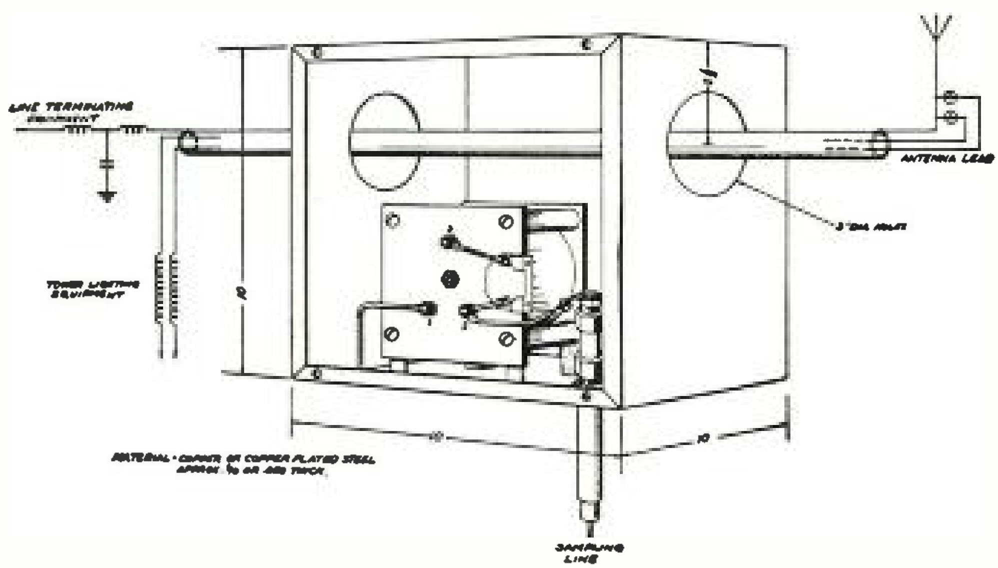


Fig. 20-9. Suggestion for housing the sampling coil, showing the position for maximum coupling.

In operation, the sampling coil should be carefully tuned to the transmitter frequency. For frequencies above approximately 1,000 kc, *L1* and *C1* will tune to resonance. Below 1,000 kc, a fixed capacitor (Faradon Model "NF") should be connected in parallel with *C1* (across terminals 2 and 3). The values for this capacitor for several frequencies ranging downward from 1,000 kc, are given below:

100 $\mu\mu$f to 800 kc
200 $\mu\mu$f to 650 kc
300 $\mu\mu$f to 600 kc
400 $\mu\mu$f to 550 kc.

A thermogalvanometer connected across terminals 1 and 2 makes an excellent resonance indicator. The coil should be loosely coupled to the antenna lead or to one of the low power stages of the transmitter, and *C1* adjusted for maximum current indication and then locked securely in place. Any appreciable deviation from true resonance will introduce error in measurements (Paragraph 1a).

For satisfactory results, care must be taken in the placement of the pickup coil. A position should be chosen to eliminate magnetic coupling to all sources except the one which is to be measured. If more than one antenna element is to be monitored, all coils should be placed in the same relative physical location with respect to the antenna leads to which they are coupled, otherwise an 180° error may be introduced. A suggestion for mounting the sampling coil is given in Fig. 20-9, which shows the position for maximum coupling. Pickup may be reduced by slightly rotating the coil assembly in the horizontal plane. Ninety degrees of rotation reduces the induced voltage to essentially zero. In cases where the current in the antenna lead is too low to give sufficient output voltage from the coil, the spacing between the coil and the antenna lead may be decreased to increase the output. But in no case should the clearance be less than one inch, in order that danger from voltage surges be minimized. In extreme cases the antenna lead can be formed into a single-turn loop parallel to the turns of the pickup coil, and the spacing reduced to approximately one inch.

2. NONRESONANT SHIELDED LOOP

a. An untuned pickup loop is easy to build and usually can be fabricated on the site. Since it requires no tuning, an initial adjustment usually proves satisfactory for longer periods of time than can be expected with a tuned pickup loop. Also, there is little danger of phase shift being introduced in the nonresonant pickup circuit. As previously stated, phase shift in the sampling circuit can cause erroneous indications by the phase meter.

b. The nonresonant loop must be rigidly mounted, and in cases where it must be coupled to points of low current, the loop might be necessarily very large to provide sufficient pickup. Since the size of the loop limits its rigidity, it is desirable to couple it to points of relatively high current if possible, particularly in high-powered installations where high r-f voltages may be present.

c. The shielded nonresonant loop in Fig. 20-10 is of single-conductor cable which can be made rigid and rather large to provide adequate pickup. The shielding must be cut back at the end and taped so that it does not short-circuit the single-turn loop formed by it and the inner conductor. This part of the loop can be effectively weatherproofed for outdoor installation by using a ¼″ copper tubing "T"

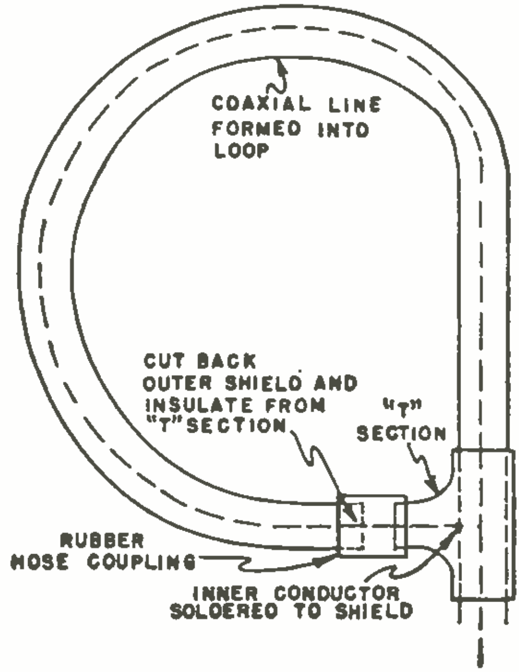


Fig. 20-10. Nonresonant single-turn loop of coaxial line which provides sufficient pickup.

section and insulating hose couplings, as suggested in the drawing. This type of loop can be pressurized along with the transmission line, if air-dielectric lines are employed.

3. LOCATION

a. Determining Factors

The ideal location for the sampling coil will vary with different installations depending upon such factors as the electrical design of the elements of the array and the feed lines, the construction of the towers or other supports, and the power used. In high-power installations, sufficient pickup is insured at many points along either the feed line or the antenna elements, but the problem of securing adequate high-voltage insulation between the sampling system and the antenna system requires consideration. In low-power installations, the desired type of sampling coil may not provide sufficient pickup unless it is coupled to a point near a current maximum, which in some installations might be near the center of the antenna element, thus presenting a problem in mounting and subsequent adjustment.

b. Sampling the Feeder at the Tower Base

In installations where the tower is fed at its base, sufficient current is usually present to induce ample voltage into the sampling system.

A sampling unit which is installed at this point is easily weatherproofed and is accessible for frequent readjustment. However, a variety of stray currents are usually encountered at this point. In some installations these stray base currents may be appreciable as compared to the absolute antenna current of the system. For example, in a vertical half-wave antenna, the capacitance of the tower to ground across the base insulators may draw an r-f current comparable in magnitude to that fed into the antenna (Fig. 20-11). Moreover, the stray cur-

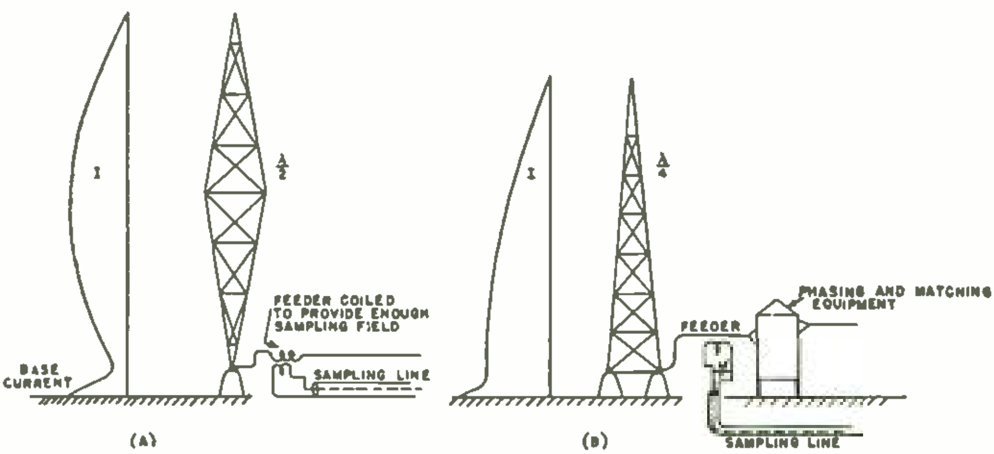


Fig. 20-11. Possible current distribution in half-wavelength and quarter-wavelength antenna systems.

rents at the base of the tower do not always provide accurate antenna-current indications under a change in operating conditions. A decrease in base feeder current might be indicated, for example, with an increase in absolute antenna current. Before making a sampling installation at the base of a tower or mast, it would seem advisable to determine both the magnitude of base currents and their relationship to the absolute antenna current under different operating conditions.

c. Sampling on the Tower Structure

A sampling system installed on the tower structure will indicate the effective value of the antenna current. It is often difficult to determine the absolute current in the tower; therefore, direct correlation between the remote meter reading and the true antenna current is difficult to achieve. An installation on the tower structure is advantageous in that ample current is usually available for sampling, and indications are not influenced by base currents if the loop is located several feet above the base. But such a system is usually costly and difficult to install. The sampling line, which can be clamped to the tower, must be brought across the tower base insulators. This requires the introduction of a high-impedance circuit at the base of the tower, which is obtained in practice by forming sufficient length of the sampling line into a coil that can be tuned by a shunt capacitor to the transmitter frequency, as shown in Fig. 20-12(A). The tuned circuit must be kept accurately tuned to the transmitter frequency, otherwise the sampling system will, of course, disturb the electrical characteristics of the antenna and will produce inaccurate phase and current relationships. In some cases, the sampling line can be spaced from the tower by high-voltage insulators, and brought across the base without an isolating network.

d. Sampling from an Adjacent Mast

To overcome the difficulty of bringing the sampling line across the base of the tower, the sampling coil can be mounted at the top of a special mast erected adjacent to the antenna tower, as shown in Fig. 20-12(B). If the antenna tower is one-quarter wavelength at the frequency used, the sampling unit at the top of a short mast erected close to the tower will provide ample voltage for antenna current and phase indications. Moreover, these indications will not be influenced by stray currents existing at the base of the tower. The sampling line can be attached to the mast and its outer shield grounded at the base. Care must be exercised in the location of the mast so that stray fields from other adjacent towers do not induce appreciable current in the sampling system. The principal disadvantages in this system are mechanical.

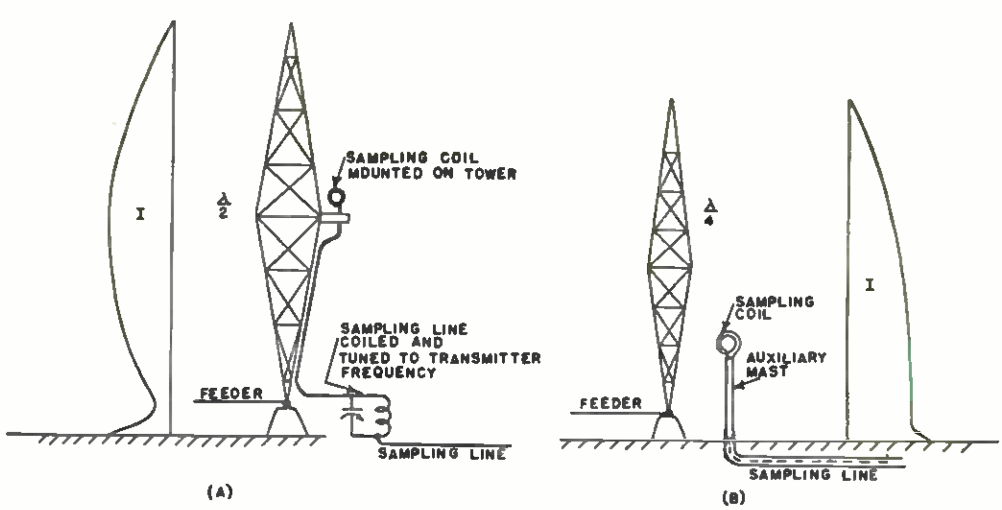


Fig. 20-12. Sampling near the point of maximum current in the tower.

The auxiliary mast must remain rigid under the most adverse weather conditions to prevent variation in the spacing between the pickup coil and the antenna tower. If the mast is laterally supported by the tower structure, the supports must be adequately insulated from the mast, which might prove too costly in some installations. In order to obtain the desired current in the sampling system, some provision must be made at the top of the mast for adjusting the position of the sampling coil and perhaps its distance from the antenna tower. In either case, where the sampling line is mounted near the antenna structure, the effect of the line on the electrical characteristics of the antenna system must be carefully considered.

4. SAMPLING LINES

Current induced in the sampling coil by the field surrounding the antenna or its feeder is fed by concentric transmission line to the remote antenna-current and phase indicators. In practice, the degree of coupling between the coil and antenna or feeder is usually adjusted so that the pointer deflection on the scale of the antenna-current indicator is identical to that for the respective antenna ammeter. This is described in section 3.

The sampling line can be any one of a number of types of concentric lines with surge impedances from approximately 50 to 100 ohms. In general, open-wire lines prove unsatisfactory; if used in the vicinity of the antenna, objectionable currents will be induced in the lines. The beaded coaxial line is an entirely satisfactory type for sampling. This type line can be obtained with surge impedances ranging from 72 to 150 ohms. Its construction provides an efficient, low-loss transfer of energy and makes it suitable for long periods of outdoor use. An ideal sampling line installation would be the use of beaded coaxial line installed within gas-filled copper tubing. Such a line could be depended upon to give reliable service over long periods of time. Solid-dielectric coaxial lines have been developed which should give long trouble-free service. They require no pressurizing.

In order that no phase shift is introduced in the sampling system, particularly if a phase meter is used, each sampling line must be terminated in its characteristic impedance, which is nominally 79 ohms for ¼ inch concentric line. The terminating networks for each line on the RCA WM-30A Phase Monitor and in the Remote Antenna-Current Indicator are identical, except that when the two units are used together the input impedance of the Phase Monitor, an effective

2,000 ohms, is shunted across the output of the Remote Antenna-Current Indicator as shown in Fig. 20-7. This should be considered when calculating the correct value for the plug-in resistors *(R4, R5,* and *R6)* of the Remote Antenna-Current Indicator. For example, assuming that a transmission line of 70 ohms impedance is used, then the value of plug-in resistor *(R)* can be determined as 68 ohms by using the formula for d-c resistance

$$Z_{\text{input}} = R_m + \frac{R_s R}{R + R_s}$$

where Z_{input} = desired input resistance

R_m = resistance of the meter and its shunt (4 ohms)

R_s = shunt resistance of Phase Monitor (2,000 ohms).

Using these values and solving for R:

$$R = \frac{2{,}000\,Z - 8{,}000}{2{,}004 - Z}.$$

As previously stated, an exact line match is unnecessary. Except at critical line lengths, variations of as much as ten per cent from the correct load value will introduce no appreciable error in either current or phase indication. Whether any error is being introduced by the Remote Antenna-Current Indicator can be determined by eliminating the unit from the circuit and calibrating the Phase Monitor. Any existing error will then be indicated by the Phase Monitor when the Remote Antenna-Current Indicator is again inserted into the circuit.

Directional Antenna Systems

Directional arrays must be used in a great majority of present-day broadcast installations. In most instances, they must be used to protect the service area of other broadcasting stations on the same or adjacent channels. Such a system also increases the service area of a station by causing the carrier wave to be reinforced in the direction of the densely populated area intended to be served. Still another application is the elimination of multiple ownership problems (anti-monopoly rulings of FCC) by using directional antenna patterns of the transmitters owned by a common company or corporation.

The design of directional antenna systems is a specialized field requiring considerable mathematics and experience. Just as in the case of transmitter site preliminary tests, this work is undertaken by licensed consulting engineers who specialize in such work. The discussion here is to eliminate the usual elements of mystery surrounding directional systems for the average operator and technician.

Conditions Governing Number of Towers. It will be obvious to most readers that directional arrays vary from two towers to as many as six as a common rule, and in some rare cases even more. The radiation pattern must be molded into a shape which results in adequate coverage of intended service areas from the necessary location of the transmitter, and at the same time provide the required protection to other stations. Economics obviously dictate a need to accomplish the results with a minimum of towers and associated phasing equipment. Under the more severe requirements, the number of antennas must be increased until sufficient control of the radiated energy is achieved to result in the necessary critical shape.

A broadly flexible rule, but one that is fundamental and serves as a starting point, is that two stations may be given the required protection with two towers. Three stations may be adequately protected with three towers, or in some cases of less severe requirements, only two towers will serve. It is a safe rule to understand that four towers will provide control of four nulls for four stations. Again, however, this job is often accomplished with three antennas. Additional requirements entering into the over-all problem, such as needed to concentrate the energy into one or two directionals to provide adequate service in addition to protecting stations in other directions, complicate the control problem and call for additional towers.

Radiation Pattern Control Principles. The simplest directional array consists of two half-wave towers spaced a half-wavelength apart with no associated phasing equipment. This combination serves as a good starting point to illustrate the fundamentals of directional broadcast arrays.

In this example, the two towers are fed with currents of equal amplitude, in step (in phase) with each other; in other words, from a point at the transmitter output which is electrically the same. The currents in each tower then reach their maximums and minimums at the same instant, assuming the transmission-line lengths to be equal. Fig. 20-13 shows the resulting wave interactions which control the final radiation pattern.

Consider first Fig. 20-13(A). Since the towers are a half-wave apart, the wave force antenna *A* (solid line) will have traveled a half-wave by the time it reaches antenna *B*. The current in antenna *B*, (shown

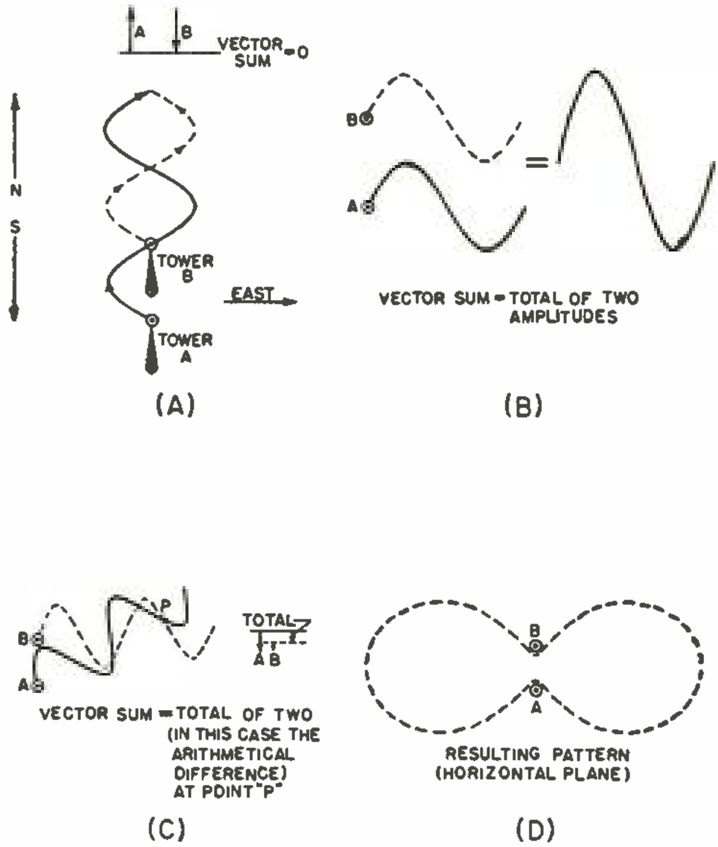


Fig. 20-13. A simple directional array consisting of two half-wave towers spaced a half-wavelength apart. (B), (C), and (D) show the wave interactions and the resulting radiation pattern.

by dotted line) will then be 180° out of phase in this direction with that of antenna *A*, and the field will tend to cancel out. Observation will show that conditions in the opposite direction in line with the towers will result in the same cancelling effect, the wave of antenna *A* now being 180° behind that of antenna *B*.

Fig. 20-13(B) illustrates the wave reinforcement that occurs in this direction. The waves of the two towers, being in phase, add to one another and the sum energy field is the total of the radiated energy from the two antennas. Obviously, the same condition holds true in the opposite direction.

Fig. 20-13(C) shows the result of the field interaction at an arbitrary point *P* at some area between those of the previous examples. The radiated waves are now slightly out of phase but not 180°, therefore, the total energy at this point is the difference of the two energies.

Fig. 20-13(D) shows the resulting directional pattern in the horizontal plane of such an array.

Consideration should now be given as to the result of feeding this two-element array with currents 180° out of phase. Fig. 20-14 shows the now obvious fact that conditions are just reversed from those of Fig. 20-13.

Fig. 20-15 illustrates a means of obtaining a cardioid radiation pattern by spacing the towers a quarter-wavelength apart with currents phased 90° apart. The reader should now be able to reason out the field interaction in this case by using the same principles set out in the previous examples.

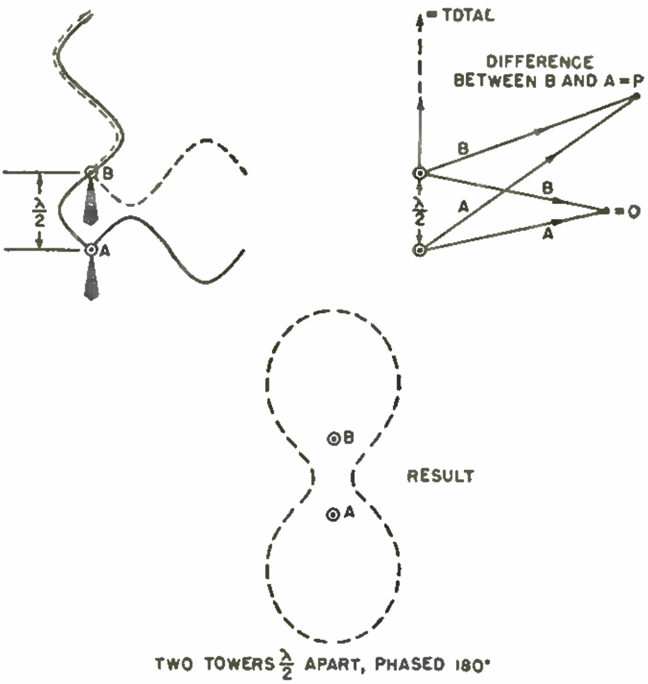


Fig. 20-14. Radiation pattern resulting when the half-wave array is fed with currents 180° out of phase. Compare with radiation pattern obtained in Fig. 20-13.

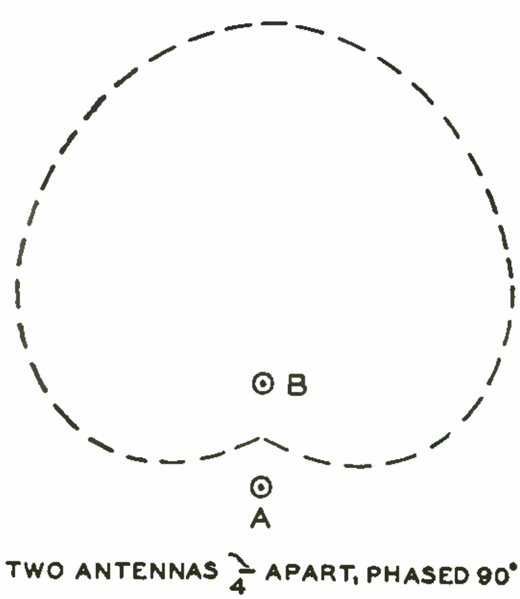


Fig. 20-15. With the towers spaced a quarter-wavelength apart and fed with currents 90° out of phase, the result is a cardioid radiation pattern.

In actual practice there are many variables involved, such as differences in amplitudes of currents, phasing of currents, spacing of the towers, etc., dictated by the requirements to be met and convenience of controlling the current magnitudes and phases. The design mathematics get very involved and complex here, and only the basic principles concern the average operating and maintenance engineer.

As a practical example, consider the two-tower directional array of Station WIRE. Nondirectional (one tower) operation is specified for daytime, with directional pattern (approx. a figure eight) necessary from local sunset time to local sunrise time. This dual operating function requires a relay switching system and is illustrated in Fig. 20-16(A), which shows the relay in position for directional operation. Fig. 20-16(A) is the schematic of the relay and phasing unit used. Fig. 20-16(B) shows the equivalent circuit for nondirectional operation, where the relay is making contact on the *A* and *B* contacts of the

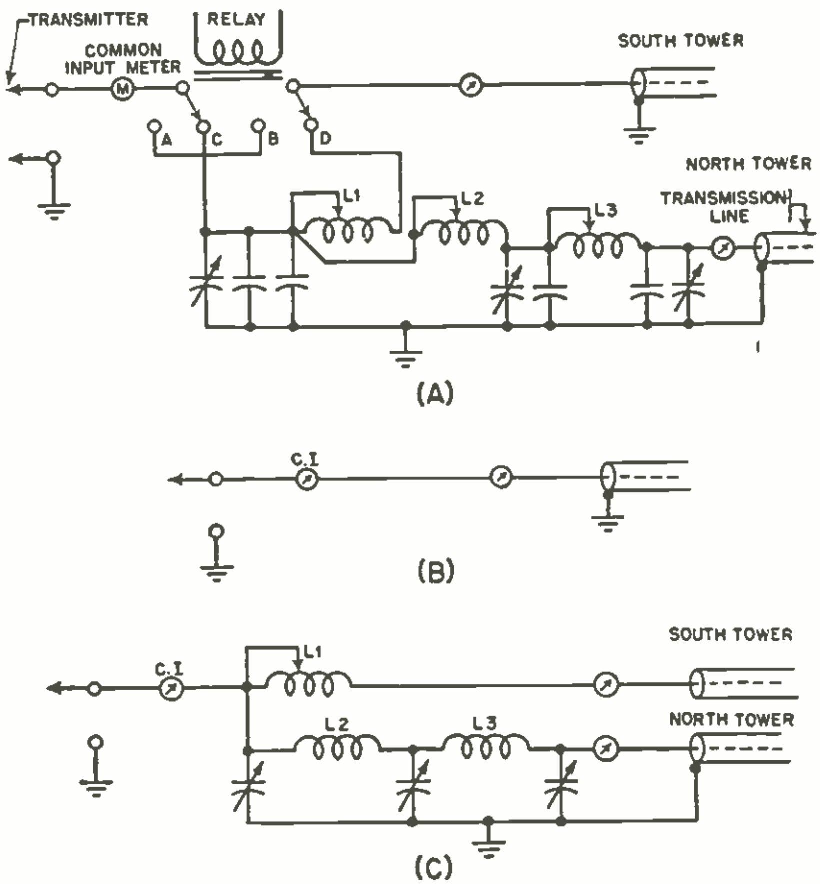


Fig. 20-16. A two-tower directional array with a dual operating function controlled by a relay switching system. The system is nondirectional in daytime and directional at night.

relay shown in Fig. 20-16(A), which disconnects the north tower and excites only the south tower. Fig. 20-16(C) shows the equivalent circuit when the directional relay is thrown to contacts *C* and *D*.

This particular directional array is defined as follows; the spacing between the towers is 351 feet (183.9°) on a line bearing 165° true. The current in the south tower leads the current in the north tower by 160.6°. The current ratio is: south tower 1; north tower 0.91. (See Fig. 20-17.)

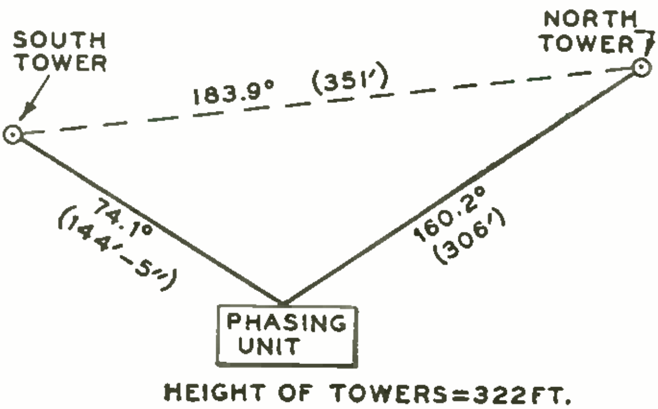


Fig. 20-17. Spacing between towers and current relations for the dual-function array of Fig. 20-16.

It should be borne in mind by the reader that a transmission line introduces a delay (in degrees) of $\theta = 360°(l/\lambda.)$ where l is the length of the line and λ is the wavelength. The phasing unit shown in Fig. 20-16 must divide the transmitter currents into two parts with the required amplitudes (power ratio) and phase angles, taking into consideration the delay occurring in the transmission lines.

In order to illustrate the function of a phasing unit, let's consider for a moment the tower array and transmission lines without the phasing control. Fig. 20-17 shows the mechanical and electrical values. The transmission line to the south tower is 144′-5″ long, or 74-1 electrical degrees. The line to the north tower is 306′, or 160.2 electrical degrees. (The operating frequency of WIRE is 1,430 kc. Hence 360°. or one wavelength is $\frac{300,000}{1,430}$; or 209 meters, or 685.5 feet, approximately.)

Therefore, the current in the south tower is delayed by 74.1° from the transmitter, and the current in the north tower is delayed by 160.2°. As conditions stand here, the current in the south tower leads that in the north tower by 160.2 − 74.1, or 86.1°.

Since the directional design factors to obtain the necessary pattern of radiation call for a current lead in the south tower of 160.6°, the phasing network must provide an additional delay in the north tower to make up this difference, or cause a phase advance in the south-tower leg of the circuit. The choice here is usually a matter of convenience, using the configuration simplest to design and control for any specified circumstance.

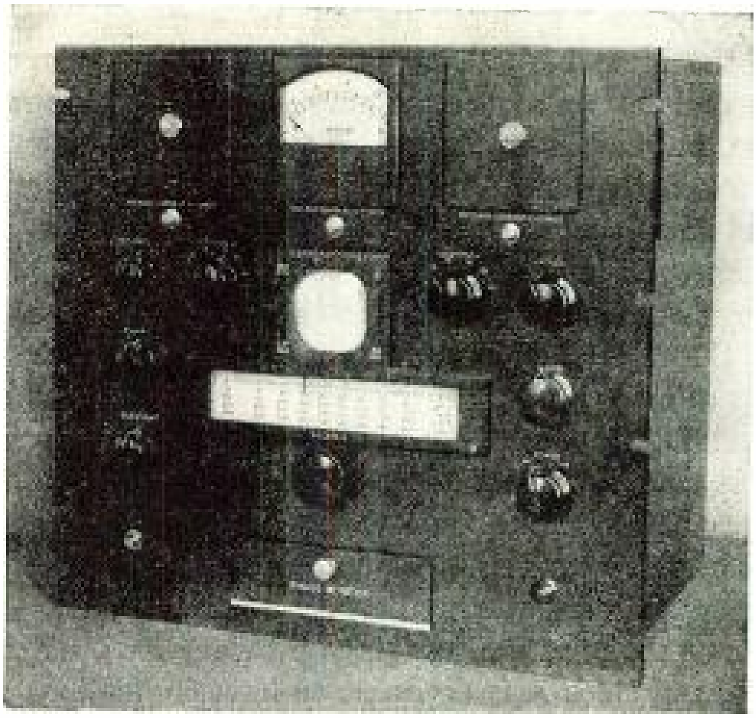
In the present example, a phase-delay circuit was chosen for the north tower to delay the current by an additional 74.5° so that the total current lead in the south tower is 160.6°.

Refer back to Fig. 20-16(A) which shows that a variable capacitor is shunted across each fixed capacitor for purposes of maintaining control after the array has been initially adjusted. These capacitors are always locked on the panel to avoid slight changes that might otherwise inadventently occur by brushing against the controls or in cleaning and maintenance routines. The FCC requires periodic checks on the directional radiation pattern and any deviation caused by changing parameters over a period of time may be corrected by this feature, along with tapped coils provided as shown.

Phase Monitoring

It is obvious from the discussion on directive arrays that it is necessary to maintain the correct relationships between phase and magnitude of currents in the different branches of this type antenna. The phase monitor is an instrument designed for the remote indication of the relative amplitude and phase of the antenna currents in the various elements of directive arrays. A typical instrument of this type is shown in Fig. 20-18.

A block diagram of the phase-measuring circuit is given in Fig. 20-19. As shown by the diagram, the "X" input and "Y" input voltages as provided by sampling lines from two towers, *A* and *B*, are fed through the "SCALE SHIFT" switch to the horizontal amplifier channel "X" and the vertical amplifier channel "Y." It should be noted that one function of the "SCALE SHIFT" switch is to reverse the "X" and "Y" channel inputs. Input to the "X" channel is fed through a phase-shifter stage which will provide in the "X" channel a 0 to 90° (maximum) phase lag behind "Y" channel depending on the setting of the main phase control ("X-LEADS-Y") on the front panel. The blanking stages in the "Y" channel operate to blank part of the trace at the proper instant, thus indicating the proper quadrant and, hence, the correct scale to be read. Operation of the phase-shifting and blanking stages is described later.



Courtesy RCA

Fig. 20-18. Panel view of a direct-reading phase monitor for use in directional antenna arrays, providing a continuous indication of antenna current phasing.

In preliminary adjustment, the input switches are placed in position to select voltage from the same sampling line, thus assuring in-phase input to both the "X" and "Y" channels. The main phase control is set on "0," to produce no phase shift. The "PHASE CAL." control, a small compensating trimmer in the "X" amplifier, is adjusted, if necessary, to make the over-all phase shift in the "X" amplifier iden-

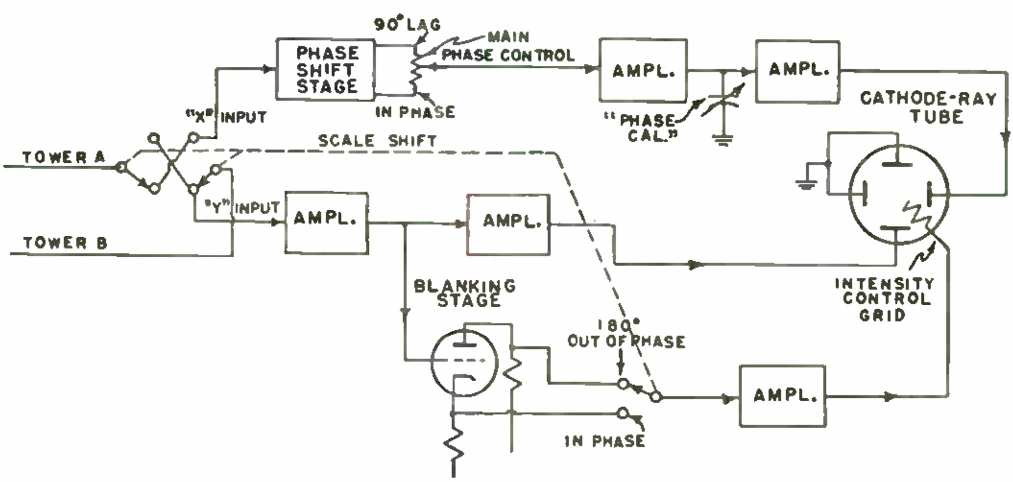


Fig. 20-19. Block diagram of the phase monitoring circuit.

tical to that in the "Y" amplifier. Then, since in-phase voltage is applied to the input of both amplifiers, the voltages applied to the horizontal- and vertical-deflection plates of the cathode-ray tube will be in phase, producing an oblique, straight-line trace on the screen. Conventional controls are provided for horizontal and vertical amplitude control and for centering of the trace. If there is a phase difference between the sampling lines to be measured, changing the input switches to these lines will then cause out-of-phase voltages to be applied to the deflection plates of the cathode-ray tube; the phase relationship of these voltages being identical to that existing between the "X" and "R" inputs under observation. At this point, a phase correction is made in the "X" channel by means of the main phase control to produce again in-phase voltages at the deflection plates, and the amount of phase correction required is indicated directly in degrees as the phase differences existing between the input lines.

Inasmuch as the phase-shifter stage provides for a maximum phase shift of 90°, and since it is desirable to have indications as given by the scale on the front panel always in terms of "X" leading "Y," provision is generally made for reversing the "X" and "Y" inputs for cases where the phase difference between the sampling lines is greater than 90° or when the phase of the voltage in the "Y" sampling line leads that in the "X" line. This is one function of the "SCALE SHIFT" switch.

Another function of the "SCALE SHIFT" switch is to shift the phase of the blanking signal 180°, providing positive determination of the quadrant in which the measured phase angle lies. Operation of the blanking stages takes place when the front panel scale-check button is pressed. A detailed description of the blanking system follows.

The Blanking Stages. As can be seen in the block diagram, Fig. 20-19, when the "SCALE SHIFT" switch is in one of its two positions, tower *A* feeds the "X" channel and tower *B* feeds the "Y" channel. Assuming that tower *A* is leading tower *B* by 180°, that no phase shift is introduced in the phase-shifting stage, and that identical phase shift is present in each amplifier, then voltages 180° out of phase will be applied to the horizontal- and vertical-deflection plates, and an oblique straight line will appear on the screen as shown in Fig. 20-20(A). The slope of the line will vary with adjustment of the gain controls, but the direction of the slope will not shift. When the scale-check button is pressed, the blanking signal applied to the control grid of the cathode-ray tube is in phase opposition to the output of channel "Y." Thus the luminous trace will be blanked during its excursion toward the top vertical-deflection plate, and slightly intensified during the opposite half of the voltage cycle, when the trace is in the lower right quadrant. The visible portion of the trace, therefore, will be in the lower right quadrant of the cathode-ray tube, as shown in Fig. 20-20(B), indicating use of the 90 to 180° scale. The phase difference as indicated by the scale pointer is then 180°. However, if the "SCALE SHIFT" switch is now placed in the other position, the voltage from tower *A* (which is leading that of tower *B*) will be applied to the "Y" channel. But the blanking signal at the cathode-ray tube, will still be in phase opposition to the tower *B* signal from channel "X" because the "SCALE SHIFT" switch, in addition to reversing the "X" and "Y" inputs, now selects output from the cathode circuit of the first blanking stage instead of its plate circuit, thereby eliminating a 180° phase shift which took place in this stage when the switch was in the normal position. Since the inputs are reversed, the blanking will now take place when the trace is in the lower right quadrant of the cathode-ray tube. The visible part of the trace is, in this case, in the upper left quadrant of the tube indicating use of the 180 to 270° scale on which the pointer again indicates 180°.

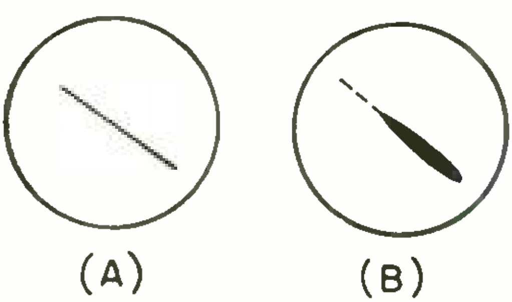


Fig. 20-20. Voltages 180° out of phase are indicated on the scope screen by a pattern as shown at (A). When the scale-check button is pressed, part of the trace is blanked out and part intensified as in (B).

The Phase-Shifter Stage. In the examples just cited, no phase shift has been introduced in the phase-shifter stage, and since the voltages were assumed to be exactly in phase opposition at the "X" and "Y" inputs and also at the deflection plates, a straight line was obtained on the cathode-ray-tube screen in each case. It is well known that when the phase of sinusoidal voltages applied to the horizontal- and vertical-deflection plates differs by angles other than 180°, a variety of elliptical patterns will be obtained depending on the phase angle. Assume now that, in the block diagram, tower *A* is leading tower *B* by 90°. The "SCALE SHIFT" switch is in position so that tower *A* feeds the "X" channel and tower *B* the "Y" channel: no phase lag is being introduced by the phase-shifter stage (main phase control at "0") and initial correction has been made for phase difference in the amplifiers by adjustment of the "PHASE CAL." control. The voltage applied to the horizontal-deflection plates, therefore, will lead by 90° the voltages applied to the vertical-deflection plates, and the pattern observed on the screen will be nearly circular. In order to measure the phase difference in this case, it is necessary to adjust the main phase control, which will introduce a phase lag in the "X" amplifier, until the circular pattern becomes a straight line. The amount of phase lag introduced represents the phase difference between the two sampling lines and is indicated directly in degrees on the scale. The proper scale to be read (0 to 90° in this case) is indicated by the quadrant (lower right) in which the pattern appears when the scale-check button is pressed. For cases where "X" leads "Y" by an angle greater than 90°, or when "Y" leads "X," measurements must be made with the "SCALE SHIFT" switch in its other position.

In making measurements, no confusion can result as to the correct position for the "SCALE SHIFT" switch for different phase angles of lead and lag between "X" and "Y." Usually a straight-line pattern can be obtained for only one position of the switch. If the straight line occurs when the phase shift introduced by the phase-shifter stage is either 0 or 90°, the straight-line pattern can be obtained in either of the two positions of the scale switch. In any event the scale-check feature determines which scale should be read.

Outline of Transmitter Installations

Modern broadcast transmitters come in sections with nearly all the parts already mounted in place. Terminal boards with numbered connections are used to provide connection to the power lines and interunit connections, with wires run in raceways behind the units or in the wiring channel along the elevated back base. Manufacturers always furnish detailed blueprints and pictures showing wire connections to each number on each unit and recommended wire size.

Audio-frequency wires should be installed with twisted-pair, leaded covered wire, and audio grounds should be separate from power line and r-f grounds. These precautions help to prevent background noise in audio circuits which would result from voltage drops along common ground wires carrying power and r-f ground currents. Audio line terminations, pads, equalizers, line amplifiers, and measuring equipment, such as frequency monitors, modulation monitors, etc., are placed in a rack apart from the regular transmitter, with power supply and audio circuit wiring run in conduits and terminated at the base of the equipment.

Before final testing can take place in a completed broadcast installation, it is necessary that a dummy antenna of the required impedance and power dissipation rating be available, or that the transmission lines and antenna circuits be adjusted properly so that the power amplifier will work into its intended load.

Provision should be made for installing external connections before the equipment is set in place. The room should have a free circulation of clean dry air. It should have: adequate illumination, both natural and artificial, provision for incoming power supply lines and broadcast audio lines including direct talk-back to studio, accessibility of good ground connection, transmission-line entrance facilities, and maintenance provisions.

Plans should be drawn up so as to permit shortest and most direct interconnection wiring. Plan carefully the following orientation of equipment; plate transformer, control console, test-speech equipment, and monitoring racks.

Some building codes will permit the plate transformer to be located inside the building, directly behind the transmitter. In this case, since it is desirable to have a clear walkway along the full length of the equipment, ample room must be provided between the rear wall and transmitter compartments. Oil-filled transformers are usually mounted outdoors with a suitable shelter, or high-voltage outdoor-type pot-heads are provided for the primary and secondary cables.

Note. PLANS MUST BE SUBMITTED TO THE LOCAL CODE AUTHORITY AND THE BOARD OF FIRE UNDERWRITERS FOR APPROVAL.

Most modern transmitters employ wire ducts for interconnecting wiring, thus eliminating need for conduit between compartments. Conduit must be planned, of course, for interconnecting the control console and test-speech-monitoring equipment racks. Wire trenches are used for long wiring runs in control and power circuits.

Most engineers of experience prefer to keep separate grounds for the antenna system and the transmitter, especially where much separation is required. This minimizes circulating ground currents. A grounding pit, using marble dust and charcoal, is often used for the transmitter grounding system. This combination sets up a chemical action that tends to maintain a clean surface on the grounding plates.

If the layout has been well planned, unnecessary crossover of wires as they enter and leave the trench are avoided. Lead-sheathed wire should be skinned back from the end at least 3 inches, and cabled together in rectangular form. Fig. 20-21 illustrates a neatly formed cable of lead-sheathed wire to a power control terminal block. They should be spotted together with solder at intervals along the run, and then well grounded at the end nearest the station grounding system.

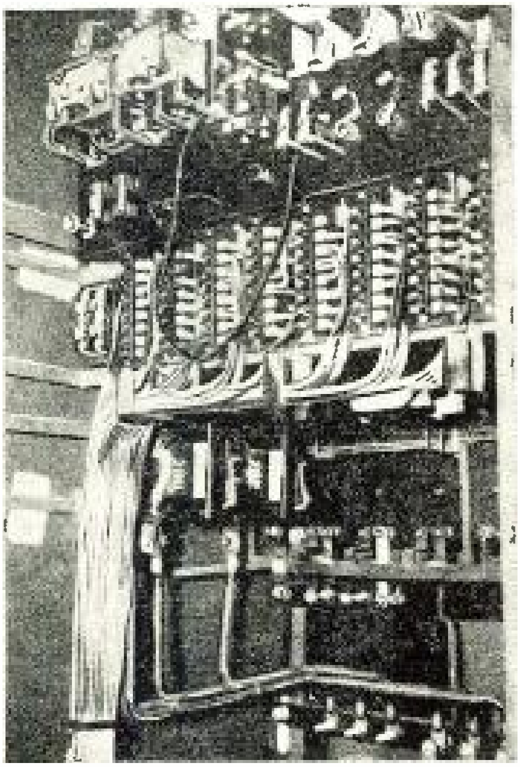


Fig. 20-21. A neatly formed cable of lead-sheathed wire to a power control terminal block.

When wiring has been completed, all safety interlocks and protective control circuits should be tested for correct operation. See Chapter 15 for a detailed analysis of control circuits.

Caution. At any new installation, acquire the habit of using the grounding stick across the high-voltage circuit before entering the transmitter for adjustments.

Circuit Tests

High power should not be applied to the transmission line until the coupling and tuning units have been adjusted to the correct characteristics by means of the oscillator as just described. High power applied to a transmission line that may be out of correct adjustment is apt to cause high standing waves on the line causing arc-overs, especially in closely spaced elements of a concentric line. When the proper impedance matching has been achieved, low power is applied to the line, and a final check made on the antenna installation by inserting an ammeter in series with each end of the transmission line. It is usually considered a satisfactory adjustment if the two meters show an indication within 20% of the value of the following formula

$$I_L = \sqrt{\frac{W}{Z_o}}$$

where I_L = transmission-line current in amperes

W = power in the radiator (antenna current squared $\times$ antenna resistance)

Z_o = characteristic impedance of transmission line.

Before applying the power to the modulation and final stages, the associated overload relays should be checked to assure satisfactory operation. This may be done by applying a low d-c voltage of approximately 10 volts between the center tap and ground of the filament transformer secondaries of the circuit tested. This will cause sufficient overload current to flow to operate the relays.

The final stage may then be tested by applying low power to the stage with the modulator power opened so that no power is being applied to the modulators. The tank circuit is then adjusted to resonance by the usual procedure. If everything is still normal, the high power may then be applied. Checking of correct neutralization to assure that no spurious oscillation exists may be made by removing one of the crystals from the spare crystal circuit so that the oscillator selector switch may be thrown to this circuit to kill the oscillator circuit with low power applied to the final stage. This should cause all grid currents to drop to zero.

The final stage plate supply should then be opened at a convenient point and power applied to the modulator plates and the static plate currents adjusted by the means provided. If trouble is experienced in bringing the static plate current down, it may be that the inverse feedback circuit is improperly phased. This is, of course, easily determined by reversing the connections of the feedback circuit and observing the effect on the modulator plate current.

Factors Affecting Hum and Noise

Noise and distortion measurements as outlined in Part 4 of this Handbook are a very important part of the transmitter personnel's duties. Hum and noise in transmitters is most commonly traced to the following factors:

1. Grid excitation to the modulated amplifier
2. Filament balancing resistors
3. Phase-balancing resistors for tubes using split filament construction of 90° phase relation to reduce effect of a-c filament supplies.

When a noise-measuring meter is used, these resistors should be adjusted to achieve minimum balance of the 60- and 120-cps components.

In order to achieve the lowest noise and distortion factor in a transmitter, the following points should be observed:

1. Correct filament and plate voltages
2. Sufficient grid excitation to modulated stage
3. Accurate neutralization
4. Correct grid-leak bypass capacitance on modulated stage.

TRANSMISSION LINES

The transmission line connects the transmitter output to the antenna or antenna tuning unit. Details of electrical design characteristics are well covered in existing literature and are not important to this text. The important features to be considered here are mechanical construction to understand installation problems, choice of proper line to meet requirements in: power handling capability, allowable power loss, proper impedance match, and economic factors. The difference in lines suitable for a-m and f-m service will be discussed.

There are two general types of radio-frequency transmission lines; the coaxial type of air-dielectric or solid-dielectric construction, and the balanced or unbalanced 4-, 5- or 6-wire open line. F-m services use coaxial lines exclusively.

Coaxial Transmission Lines

The coaxial line uses inner and outer conductors made of rigid or semiflexible copper tubing, with either air as the dielectric, or a solid dielectric of polyethylene. Solid-dielectric cables, while having the advantage of flexibility and comparative ease of installation, are less efficient in transmission of r-f energy (particularly at uhf and vhf), and have less power carrying capacity. More on this later.

Fig. 20-22 shows the construction of a large-diameter line using cross pin insulators to support and space the inner conductor. Insulators of Steatite are spaced throughout the length of the line at intervals depending upon design. Close spacings help to make the line more uniform, and are desirable for constant impedance characteristics. Longer spacings, however, minimize insulator loss and make for higher relative efficiency. In practice, the insulator loss is minimized by using insulators as small as possible considering mechanical strength. At standard broadcast frequencies, insulator spacing is not extremely important and is usually dictated by mechanical considerations alone. At the higher frequencies used by f-m and tv broadcasters, spacing is usually made so that the impedance at the highest operating frequencies varies no more than 3% from the lower frequency (standard broadcast band) values. This practice has resulted in an average spacing value of 12 inches for f-m and tv lines.

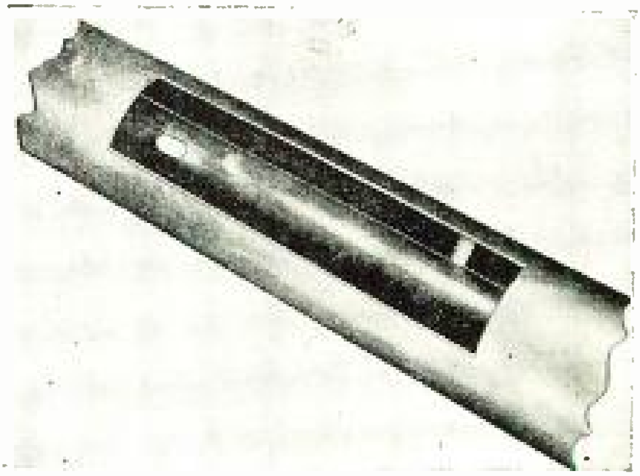


Fig. 20-22. Line using cross pin insulators to support and space the inner conductors.

Courtesy Andrew Corp.

Impedance Values of Coaxial Lines. It is now universal to use 70-ohm coaxial lines for standard broadcast transmitters, and 51.5-ohm lines for f-m transmitters. This holds true whether using air-dielectric or solid-dielectric cables. It is possible and practical in many cases to connect the rigid-type line to a solid-type cable without matching sections, by means of suitable connectors that may be obtained.

Power Handling Capacity. The maximum power which a coaxial cable will carry is primarily limited by temperature rise in the line when the r.f. is applied. At standard broadcast frequencies, voltage breakdown is the limiting factor. Above approximately 50 Mc, a temperature rise of 40° C. in the outer conductor is considered to be the point of maximum power value. Various conditions, however, aside from the actual r-f power applied, may affect temperature rise in the line. The amount of ventilation, the nature of any type of enclosure and, most important, the standing-wave ratio, are all contributing factors.

In practice, it is customary to consider the probable standing-wave ratio to be encountered (considered later), then divide the maximum power rating of the line by this ratio. For example, if a standing-wave ratio is 1.5, and a 5,000-watt rated line is considered, the maximum power recommended to be fed into the line is $\frac{5{,}000}{1.5}$ or 3,333 watts.

Table 6

AVERAGE MAXIMUM POWER RATINGS OF AIR-DIELECTRIC COAXIAL LINES

Size (inches)	Average Maximum Power Ratings (watts)
3/8	500
7/8	3,000
1 5/8	12,000
3 1/8	50,000
6 1/8	150,000

There are four standard diameters of rigid (air-dielectric) coaxial lines as follows: 7/8, 1 5/8, 3 1/8, and 6 1/8 inches. Most manufactuers in the field design equipment to accommodate these standards sizes only. In addition there is a 3/8-inch line used in some very low power transmitter installations.

Choosing a Coaxial Line. The choice of impedance value is, of course, dictated by application. Choose a 70-ohm value for a.m. (when coaxial is used), a 51.5-ohm line for f.m.

The choice of size of line depends upon maximum power to be applied, and maximum allowable attenuation. Table 6 gives the average maximum power ratings, based on possible flashovers, for the standard broadcast band. It should be remembered that for a-m service, the maximum power occurs at 100% modulation, and is more than the rated power output of the transmitter. Except when lines are very short, they should not be operated at this maximum power rating.

Choice of a coaxial cable for f-m broadcast services is somewhat more involved. The procedure may be outlined briefly.

1. *To find the power required at the antenna terminals,* divide the ERP (effective radiated power) by the power gain of the antenna to be used.

2. *To find the minimum allowable transmission-line efficiency,* divide the antenna input power from 1. above, by the maximum power output of the transmitter.

3. Choose a line from one of the standard sizes which has an efficiency equal to or greater than the minimum allowable value of 2. above. See Fig. 20-23.

4. *To check the necessary transmitter output power,* divide the antenna input power obtained in 1. above by efficiency of the line selected in 3. above.

5. Check the power rating of the cable selected from 3. above. Should this rating be less than the necessary transmitter output power from 4. above, a larger line must be used.

Table 7

AVERAGE POWER RATINGS OF STANDARD SIZE LINES

Size (inches)	Maximum Power in F-M Broadcast Band (Unity Power Factor) (watts)	Maximum Power with 1.75 Voltage Standing-Wave Ratio (watts)
3/8	1,370	772
7/8	3,000	1,710
1 5/8	10,000	5,700
3 1/8	42,000	24,000
6 1/8	166,000	95,000

As a practical solution of the above procedure, assume an ERP of 33 kw to be obtained. A 10-kw transmitter will be installed, with an Andrews 4-bay f-m antenna of 3.7 power gain (discussed elsewhere in this chapter). A line length of 200 feet will be necessary. These values are used in the above steps as follows:

1. ERP of 33 kw divided by the antenna power gain of 3.7 = 8,920 watts power required at the antenna terminals.

2. 8,920 watts divided by 10,000 watts = 0.892 or 89%.

3. Observation of Fig. 20-23 shows that the smallest line suitable is a 1 5/8 inch diameter, for a length of 200 feet.

4. 8,920 watts divided by 0.892 = 10 kw.

5. Observation of Table 7 of power ratings shows that the 1 5/8 inch line is rated at 10 kw at unity power factor. As discussed previously, this power rating must be divided by the probable standing-wave ratio (voltage) to be perfectly safe. The Andrews antenna has a VSWR of less than 1.4, but it is always good practice to assume the maximum allowable VSWR of 1.75 in our computations. The

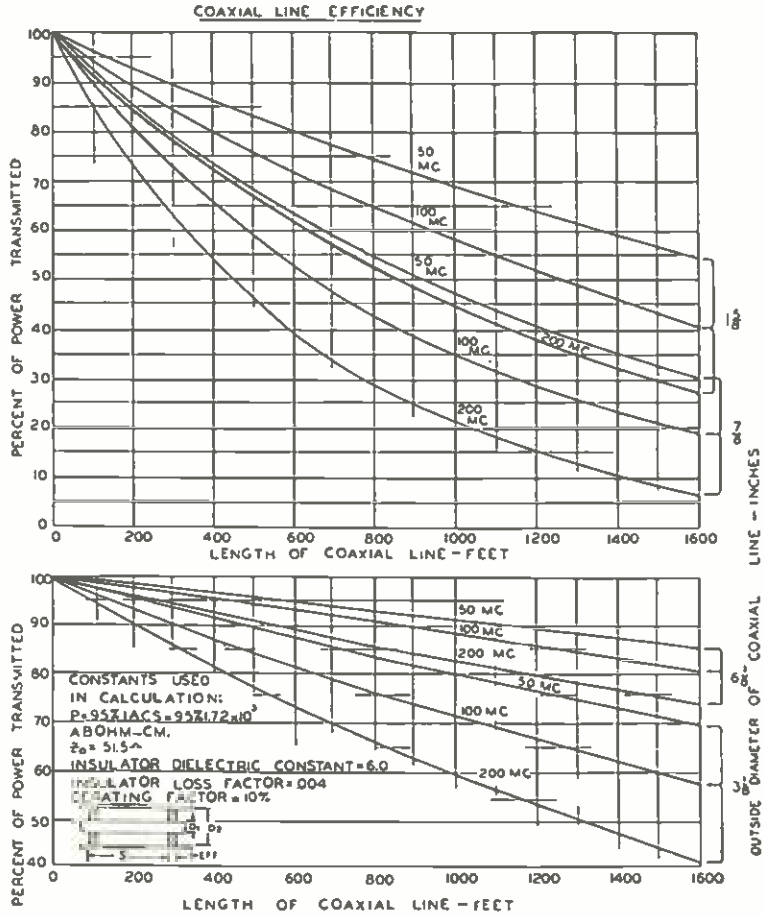
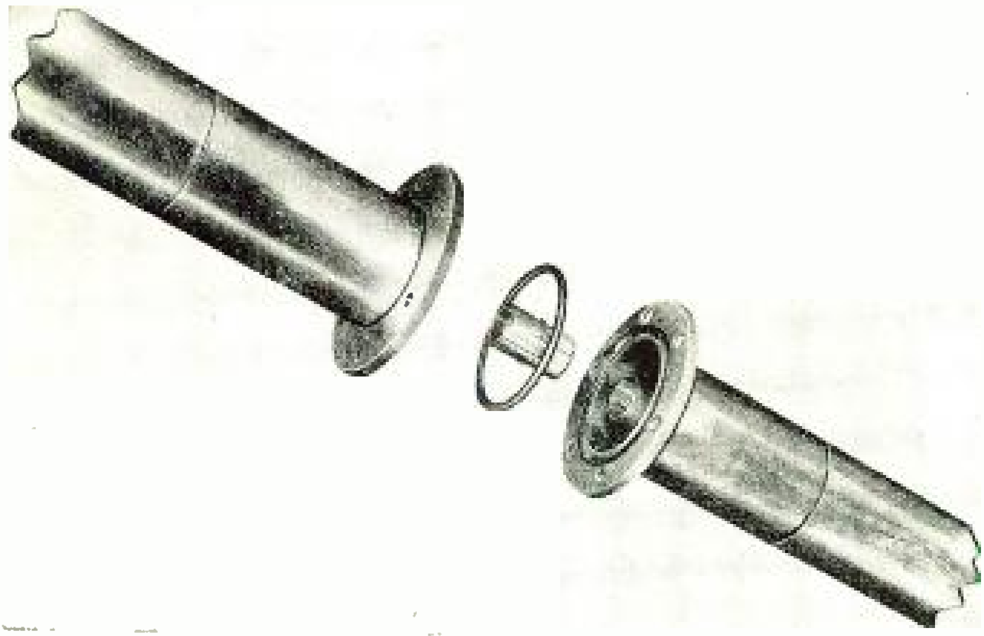


Fig. 20-23. Relation of length of coaxial line to percent of power transmitted for different frequencies and line diameters.

average power rating of the 1⅝ inch line with 1.75 VSWR is only 5,700 watts. Therefore, we must look to the next larger line, the 3⅛ inch, which is observed to have a power capacity of 24 kw at a VSWR of 1.75. This, then, would be the correct choice for this example. The higher efficiency of this line must then be considered in the computations and it will be found that a lower transmitter output will be required.

Installation of Coaxial Lines. Semiflexible soft copper lines of the ⅜ inch size are usually manufactured in 100-foot lengths which are spliced together by silver brazing to obtain a gastight and electrically secure connection. They are crated in coils, usually cut to the exact length specified. This type line is often used in larger stations for r-f sampling lines such as remote antenna current meters and directional phasing monitors.

Rigid lines of hard-temper copper tubing are shipped in 20-foot straight lengths. Flanges are usually silver-soldered on both ends of each length at the factory. Fig. 20-24 illustrates the physical construction of the coaxial lines. It is seen that flanges are provided with bolt holes so that the lengths of line may be bolted together. Grooves in the faces of the flanges accommodate the round gasket shown (known as an O ring) which makes the connection gas tight and weatherproof. The inner conductors are connected by the slotted *bullet* spring shown, providing a solderless connection necessary for field installation.



Courtesy Andrew Corp.

Fig. 20-24. Flanges on rigid coaxial lines. The O ring makes the connection gas tight and waterproof.

When it is necessary to change the direction of line runs, special elbows of 45 or 90° are used. Fig. 20-25 shows the Andrews 90° elbow which is equipped with a special flange in two pieces. The flange that provides the O ring groove is silver-soldered to the bend in the usual manner, while the second ring containing the bolt holes is assembled loosely on the bend. This provides a means of rotating the flange so that proper hole alignment may be obtained without drilling bolt holes to accommodate any orientation of the elbow.

It is sometimes necessary to cut a 20-foot section of line in the field. When this must be done, a device known as a clamp connector (Fig. 20-26) may be used to provide a solderless fitting which substitutes for the flange it was necessary to cut away. When cutting is necessary, it is preferable to make the cut at the midpoint between insula-

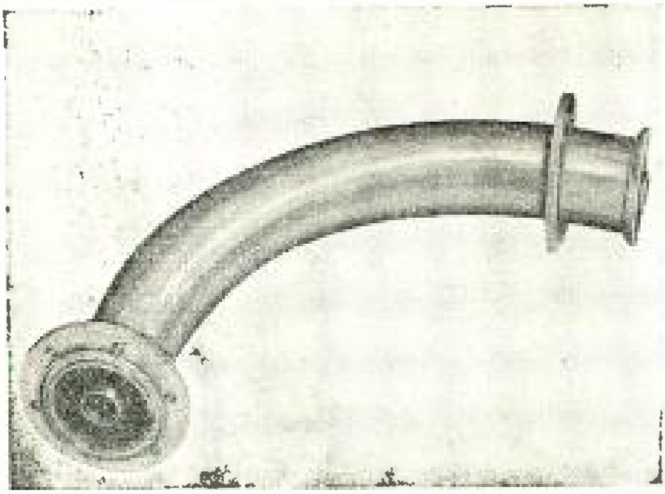


Fig. 20-25. The Andrew 90° elbow.

Courtesy Andrew Corp.

tors, since this is the point at which the characteristic impedance of the line prevails. Cutting at any other point will, therefore, somewhat disturb the impedance of the line. This practice is very important on f-m line installations where the VSWR must be kept as low as possible. If the cut is necessary very near the transmitter and rather than near the antenna end, the procedure is not so important. Midpoints between insulators are usually marked by bands on the outside of the outer conductor.

Providing for Expansion. Soft temper coaxial cable in the 3/8 or 7/8 inch size rarely need any special provision for expansion due to temperature changes, since they are somewhat flexible by nature. Rigid lines, however, must have some means of allowing for expansion and contraction. It has been found from experience that a temperature range from winter cold to summer heat in severe climates will cause

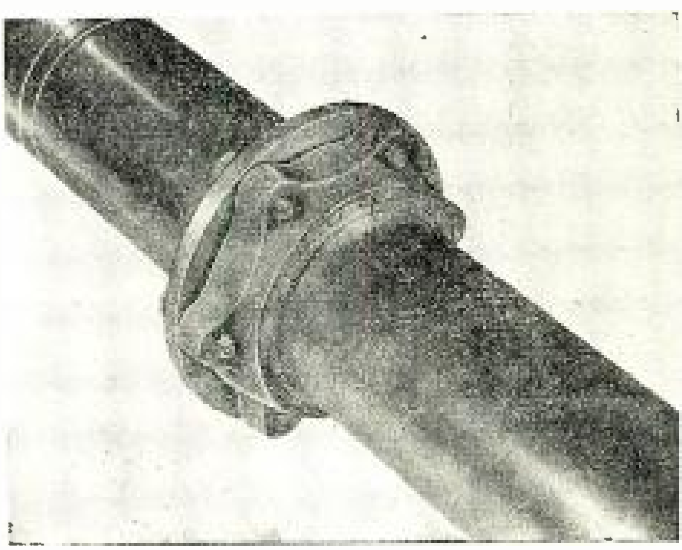


Fig. 20-26. A clamp connector used to provide a solderless fitting when a flange must be cut away.

Courtesy Andrew Corp.

an effective variation in line length of about 1¼ inches per 100 feet. Thus, for example, on a 400-foot run, provision must be allowed for a variation of around 5 inches in length.

There are two general types of expansion, one which must be considered in the design and construction of the line itself, and one which the installation engineer must provide for. These two types are known as differential expansion and over-all expansion.

Differential expansion occurs from unequal temperatures of inner and outer conductors. Under normal operating conditions, the inner conductor develops a temperature rise in relation to the outer conductor. This causes expansion of the inner conductor. Another feature causing differential expansion is that it takes considerably longer for the inner conductor to follow variations of temperature in the outer conductor due to sudden weather conditions. For example, the outer conductor may be suddenly cooled from 100° to 70° F during a summer shower. It will take the inner conductor considerably longer to equalize in the same ratio.

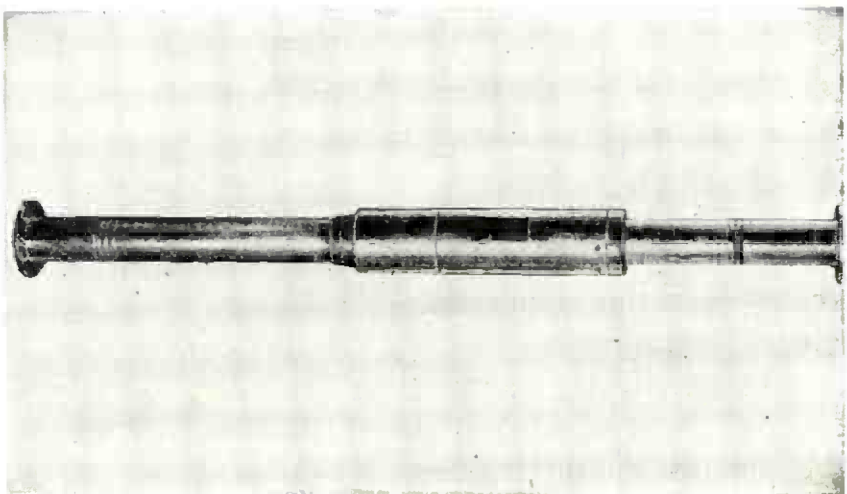
This type of expansion is provided for line construction by using an inner conductor slightly shorter than the outer. Thus when using the spring *bullet* inner connector, one end will not fit up too tightly against the shoulder, providing for its own expansion, or shrinking of the outer conductor.

Over-all expansion must be provided for in installing the line. Long horizontal runs must be given some means of relief from such expansion. When running up steel towers to feed an f-m antenna, consideration must be given the fact that the expansion coefficient of copper is as much as 50% greater than that of steel. The two metals cannot, therefore, be connected inflexibly together. It is said that several tons of pressure may be developed where no expansion provision is made, causing failure of transmission-line couplings, supports, or the tower itself.[2]

There are several alternative methods of providing for over-all expansion. When the line is so located that it may change directions several times, expansion is automatically allowed by the 45 or 90° elbows used. Support standards using roller supports will allow longitudinal motion of the line. A combination of these two provisions usually allows a sufficient safety factor for line expansion, and is widely used for long horizontal runs to the towers.

[2] Cox, C. R., "Coaxial transmission lines," *FM and Television*, Part 1, vol. 6, pp. 28-32, June, 1946; Part 2, vol. 6, pp. 30-34, July, 1946.

Such methods, however, are not entirely practical on vertical runs up a tower such as used in an f-m installation. For such applications, various types of "expansion joints" have been developed. Fig. 20-27 illustrates one type, which consists of sliding telescoping outer and inner conductors with a sliding gas-tight seal. This joint is capable of a 4-inch line displacement, entirely adequate for a 200-foot line used in temperature variations over a range of $-50°$ F to $150°$ F. Such expansion joints are often used on horizontal runs as well as vertical runs.



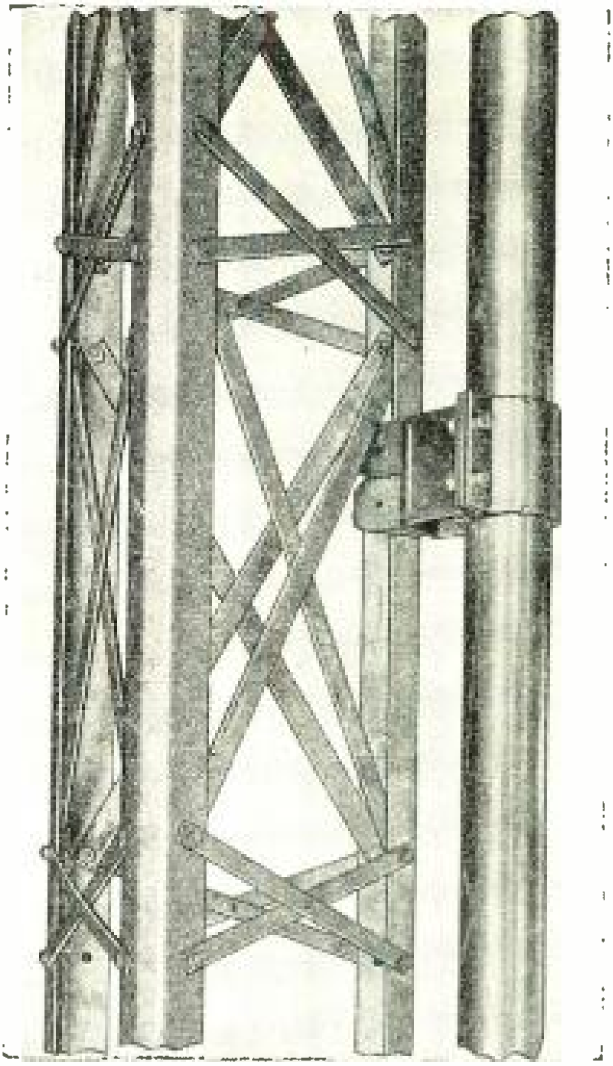
Courtesy Andrew Corp.

Fig. 20-27. A typical expansion joint used in f-m applications.

The entire line support system consists of rigid mounting clamps, support brackets or railings allowing axial movement, and sometimes one or more expansion joints. On long vertical runs of over 150 feet, expansion joints should be considered a prime requirement. On vertical runs of this type, the line is assembled starting at the bottom, the bottom section being rigidly attached to the tower as illustrated in Fig. 20-28. Additional line sections are supported at approximately 15-foot intervals preferably with sliding support brackets up to 200 feet of line. At this point an expansion joint should be installed, with rigid clamping used on the next 20-foot section. The process is then repeated for each 200 feet of line.

Support brackets may be obtained to fit practically any mounting problem encountered in the field. When it is necessary to insulate the transmission line from the tower (see below) the use of insulated support brackets such as shown in the photo of Fig. 20-29 are used.

A word of caution should be given here. Whenever it is necessary to insulate a transmission line from the tower, it should be far enough



Courtesy Andrew Corp.

Fig. 20-28. Attaching the line to the tower.

separated from the climbing ladder, where used, so that a man climbing the tower will not contact the line. Such contact could cause severe r-f burns. Towers not incorporating a climbing ladder are particularly hazardous to the climber under such circumstances and he should be cautioned about the conditions.

A-M F-M Tower Coaxial Line Procedures. When an f-m antenna must be mounted on an a-m tower, it is necessary to mount the transmission line so that the a-m electrical characteristics of the tower remain undisturbed. Several pratical methods prevail; the insulated line method and the "bazooka" construction.

1. Insulated Line Method: It is necessary in either of the methods to be described to provide isolation between the two r-f voltages by

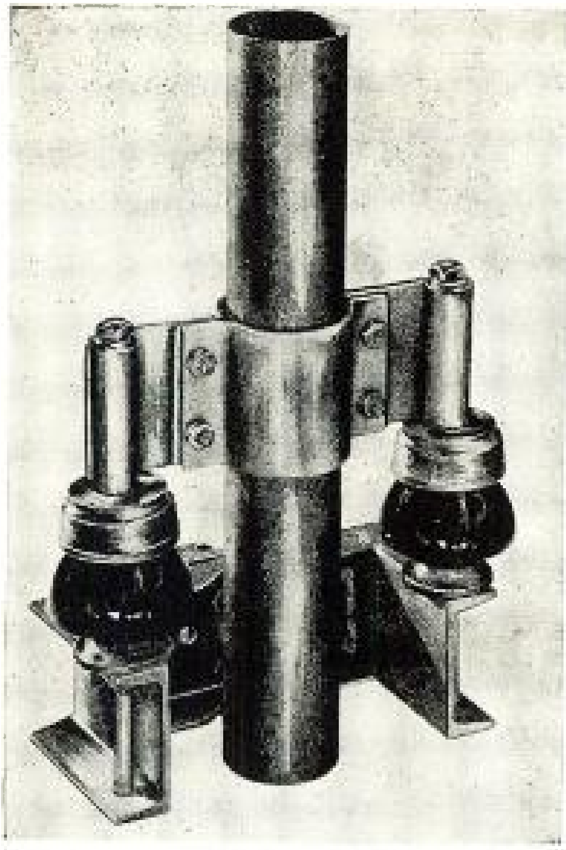


Fig. 20-29. Insulated support brackets are used to insulate the line from the tower.

Courtesy Andrew Corp.

causing a high impedance to exist between the base of the a-m tower and the outside surface of the f-m transmission line. In the first method to be described, the f-m line is insulated from the a-m tower by means described above for one-quarter wavelength (at the a-m frequency) starting at the exact base of the tower. The outer conductor is then shorted to the tower at a point one-quarter wavelength (a.m.) up from the base. Along the insulated portion of the line, the tower itself is considered to form the outer conductor. In practice, this point is determined by trial, and is only approximate. The point at which minimum detuning of the a-m tower is obtained is the optimum shorting point. Thus we have a transmission line one-quarter wavelength long with the end shorted (considering the a-m power) and a high impedance exists across the tower base insofar as the a-m power looks into the f-m line. Above the shorting point, the f-m line is attached directly to the tower.

In actual installation practice, it is advisable to short the line slightly less than a one-quarter wavelength from the tower base, and connect a variable capacitor across the "open" end at the base. (From outer conductor of line to tower.) An exact adjustment may then be made. The capacitor should be of 400-1,000 ohms reactance at the a-m frequency, with sufficient voltage rating.

2. The "Bazooka" Method: This method differs from the above only in physical construction, the electrical theory of isolation by means of one-quarter wavelength lines being retained.

A "bazooka" is a one-quarter wavelength line (at a-m frequency) assembled along the ground from the base of the tower. It actually consists of a one-quarter wavelength shield installed about a transmission line as is illustrated in Fig. 20-30. The line is insulated from the shield and the shield is grounded to eliminate radiation at the a-m frequency. The f-m line runs through the shield as the inner conductor. The line is then connected to the vertical tower throughout the entire run, eliminating the need for insulating the line on the tower. The outer shield may be either a metal hood or a wire cage consisting of six wires supported about the line by means of a metal ring with the wires run through holes around the ring.

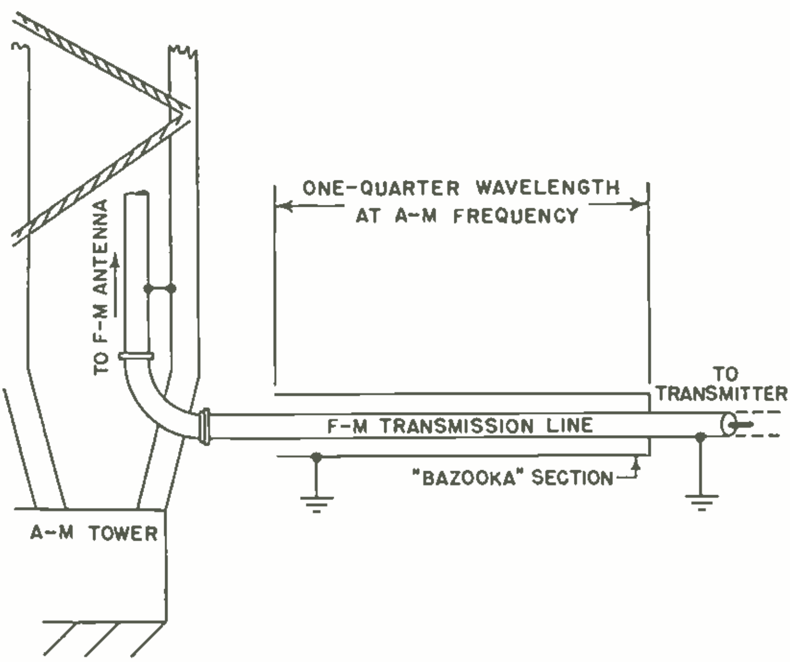


Fig. 20-30. A quarter-wavelength "bazooka" section used to isolate an f-m transmission line.

As in the first method, the "bazooka" section may be cut shorter that an actual quarter wavelength and a variable capacitor shunted across the open end to achieve exact resonance. The same reactive and voltage values prevail as in the first case.

Gassing Provisions. In air-dielectric coaxial lines it is advisable to introduce dry air or nitrogen into the line to prevent moisture condensation. This is done unless the line installed is very large in ratio to the applied r-f power.

When gassing is used, it is also necessary to use gas opening flanges at intervals on extremely long lines, or at the antenna end in order to bleed the line. This means that any moisture which does collect may be removed by allowing the dry air or nitrogen to blow through the line, evaporating the moisture. When lines are run to the top of a tower for an f-m antenna, a line is run back down from the gas seal at the antenna to the tower base, so that the gas release valve may be operated at that point.

Gas pressure is not critical, and may vary in pratice anywhere from 1 to 30 pounds.

Solid-Dielectric Coaxial. Advantages of this type line are: maximum flexibility, comparatively low initial and installation cost, no gassing required, less mounting problems since expansion is no consideration. Disadvantages are: greater attenuation per foot due to the fact that up to a certain point no further increase in diameter will lower insulation loss, less permanent in use when exposed over long periods of time to outdoor weather.

Table 8

A.M. (For 70-Ohm Applications)

Type No.	Actual Z_o	Maximum Operating Volts RMS	db Loss per 100 ft Average in Standard Broadcast Band	Maximum Total Watts Input for Standard Broadcast Frequencies
RG 11 U	75	4,000	0.115	1,800
RG 34 U	71	5,200	0.115	4,500
RG 59 U	73	2,300	0.260	860
Amphenol 21-125	72	10,000	0.020	8,000
F.M. (For 51.5-Ohm Applications)				
RG 8 U	52	4,000	2.10	880
RG 17 U	52	11,000	0.850	3,150
RG 19 U	52	14,000	0.700	4,700
RG 58 A	52	1,900	4.10	210

Solid-dielectric lines are composed of a highly resistant vinyl outer jacket, an inner conductor of stranded wire (size determined by type number), held in very close tolerance coaxially to an outer conductor of shielded braid by a polyethylene dielectric. Such lines are often used in low-power a-m and f-m installations where the run may be kept under 100 feet, and the coaxial adequately shielded from extreme weather conditions. These are also popularly used for monitoring lines such as remote antenna phase monitor, etc.

The most popular sizes of solid-dielectric coaxial used in a-m and f-m low-power transmitter installations are listed in Table 8, along with the technical specifications.

Caution: In regard to the maximum power rating, always remember to divide this rating by the maximum allowable VSWR (voltage standing-wave ratio). These values are 2 for a.m.; 1.75 for f.m. Although the actual VSWR should be lower than these values, a safety factor is allowed by this procedure.

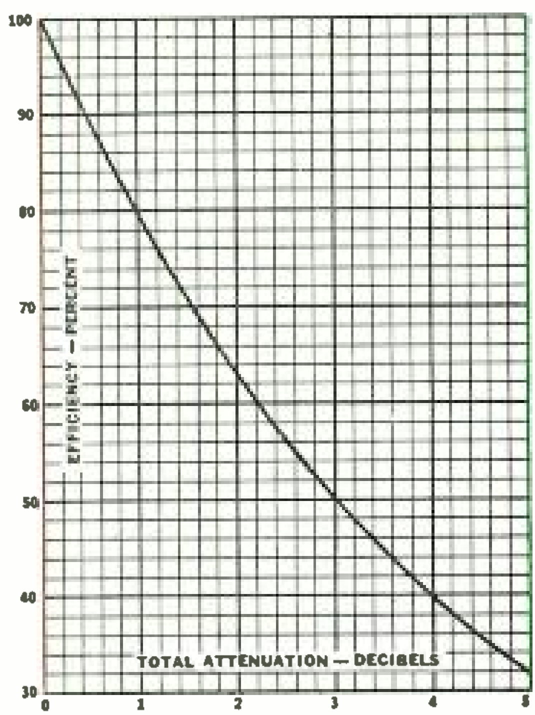


Fig. 20-31. Conversion of db attenuation to percent of efficiency.

The graph of Fig. 20-31 shows the conversion of db loss to efficiency in per cent, for those readers concerned with percentage computation as discussed previously.

Open Wire Transmission Line

The six-wire open line was originally developed by RCA, and is used to feed directional arrays as well as a single nondirectional tower. It is suitable for use over the standard broadcast band under all weather conditions. This type of line is readily serviceable, is more economical in initial cost, and, of course, does not require gassing equipment used with most coaxial lines.

Fig. 20-32 shows the principle of operation. Two central power conductors are used, surrounded by four grounded conductors. Radiation from the power conductors is made negligible by this method. The line system has a characteristic impedance of 230 ohms.

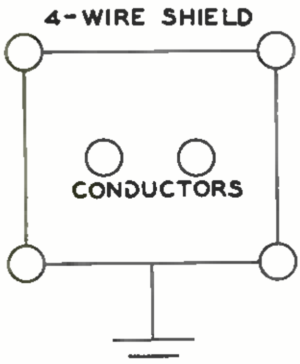


Fig. 20-32. The six-wire open line.

Fig. 20-33 shows the details of the mounting poles and line configuration. The bracket bayonet assembly mounts the station post insulator and also supports the four ground wires. The two power conductors are supported on the notches shown in the central insulator. It may be observed from Fig 20-33 that lightning protection is afforded by a horn gap, antisurge inductor, and copper tubing for ground and lead-in connections.

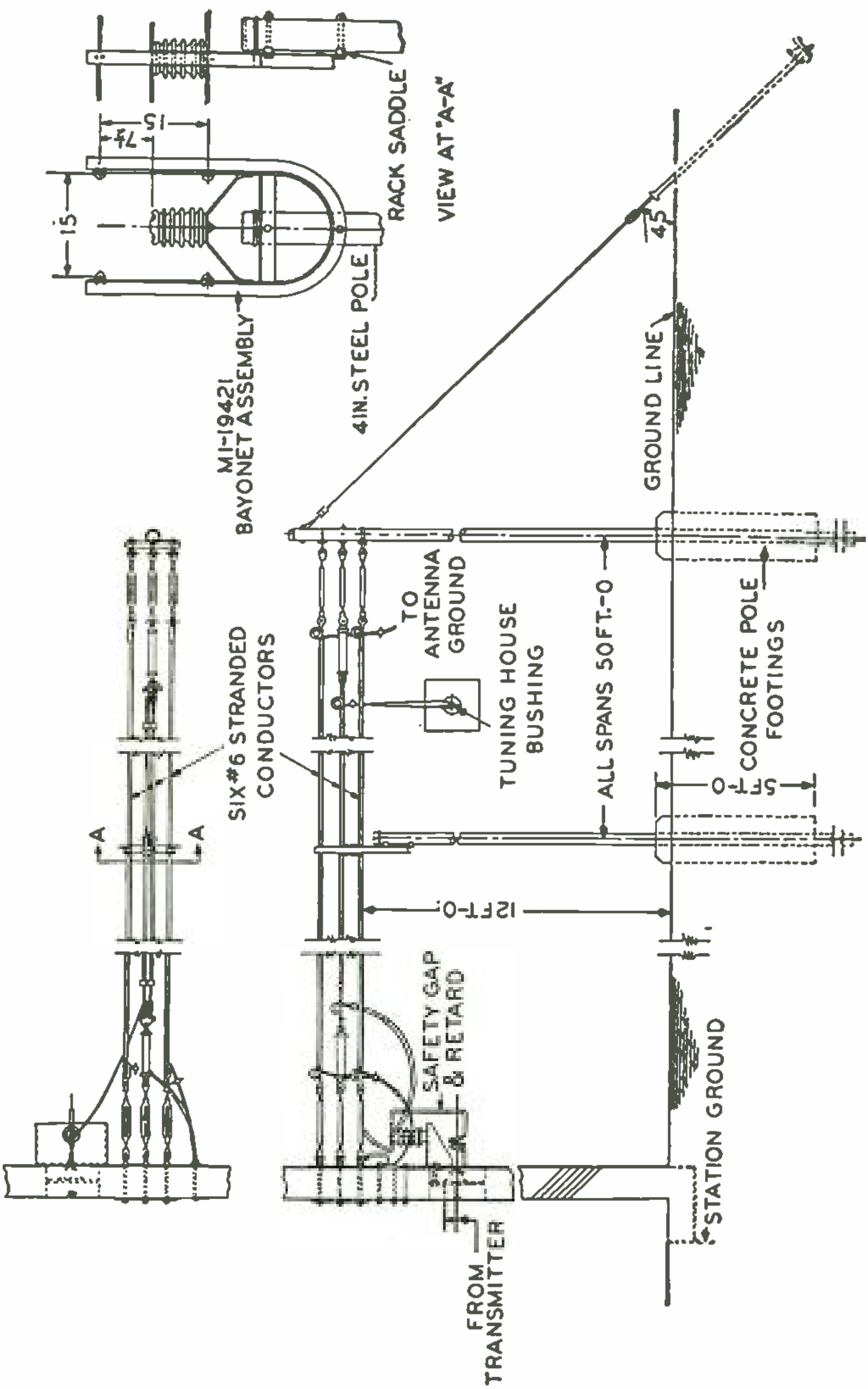
Antenna Tuning

The circuit of a single antenna tuning unit often consists of a single T-section low-pass filter as illustrated in Fig. 20-34. The two series inductors allow independent adjustment of their respective terminating impedances; *L1* for the transmission line, *L2* for the antenna circuit. The capacitive shunt leg, common to both branches, is given a fixed value determined by the operating frequency of the station as described in the following discussion.

What the Coupling Network Does. This T-section network has two primary functions:

1. To match impedances
2. To tune the antenna to the exact resonant frequency of the station.

For tuning the antenna, the coil *L2* is used to series-resonate the capacitive reactance of an antenna, or, should the reactive component be inductive, it is to be considered absorbed into the inductance *L2*.



Courtesy RCA

Fig. 20-33. Physical details of the six-wire open line.

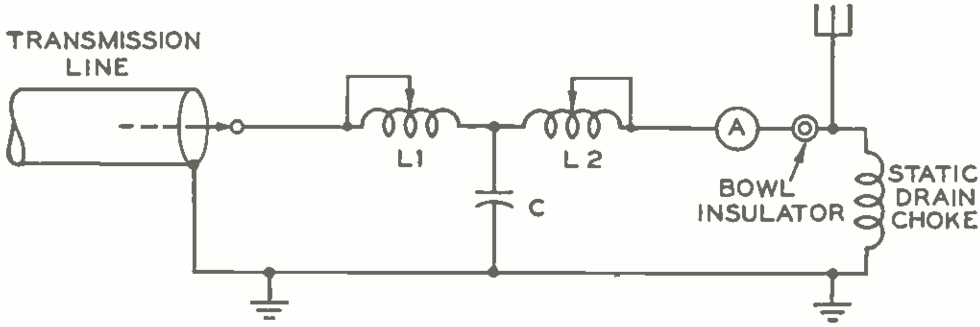


Fig. 20-34. T-section low-pass filter for single antenna tuning unit.

Coil *L1* is then used to match the resistance of the transmission line to the resistance component left in the antenna circuit.

Table 9

Antenna Height in Electrical Degrees *	Self-Supporting Type		Guyed Mast Type	
	R	jx	R	jx
50	7	—j100	8	—j222
60	9	—j70	13	—j170
70	14	—j25	19	—j75
80	20	+j11	28	—j28
90	40	+j35	36	+j0
100	60	+j80	80	+j140
110	90	+j90	140	+j320
120	175	+j80	220	+j500
130	190	+j15	370	+j600
140	165	—j70	660	+j480
150	130	—j85	1100	+j0
160	82	—j55	550	—j250
170	60	—j25	280	—j450
180	40	—j5	180	—j500
190	28	+j25	120	—j430
200	23	+j50	80	—j400

* Height in electrical degrees = Height in feet $\times$ frequency in kc $\times$ 1.016 $\times 10^{-6} \times$ 360. *Courtesy RCA.*

Table 9 presents the average resistance and reactance values of different tower heights in both self-supporting and guyed varieties, These are not absolute values, which must be accurately determined by an r-f bridge or other means as described later.

This type circuit may be seen to form a low-pass filter network with excellent harmonic attenuation. It should also be observed that the reactance values may have a phase shift between zero and $-180°$ (cutoff). In practice, a value of around $-90°$ is chosen, in which case the following formulas hold true for each reactive element:

$$X1 = +\sqrt{R1R2}$$
$$X2 = +\sqrt{R1R2}$$
$$X3 = -\sqrt{R1R2}$$

where

$R1$ = line impedance
$R2$ = antenna resistance.

There are two accepted methods of tuning a broadcast antenna; the r-f bridge method and the substitution method. Both are described in the following paragraph.

R-F Bridge Method. This is the more reliable of the two common methods, and is recommended wherever possible to obtain a good r-f bridge.

First, it is necessary to determine the impedance value of the antenna at the operating frequency, so that the proper values of the inductance and capacitance arms may be computed.

Impedance measurement of the antenna is made directly at the antenna tower input terminal with the line disconnected.

As an example, suppose we have a tower of 190 electrical degrees, with 28 ohms of resistance and $+j25$ ohms reactance (see Table 9). The transmission line is a 70-ohm standard impedance line.

Substitution of these values in the equations above give:

$$R1 = 70$$
$$R2 = 28$$
$$\sqrt{(70)\,(28)} = \sqrt{1960} = 44.3 \text{ ohms (approx.)}.$$

Therefore, $X1$, $X2$ and $X3$ should have a value of 44.3 ohms at the operating frequency.

The antenna reactance in this case is $+j25$ ohms. We assume that this reactance is a part of the value of $X2$. Therefore, we subtract the

25 ohms of positive antenna reactance from 44.3 ohms, leaving 19.3 ohms to be obtained in *X2*. When *X1* and *X3* have been adjusted to 44.3 ohms, an impedance match is effected between the transmission line and the antenna resistance. X_2, adjusted to 19.3 ohms, cancels the antenna reactance which would cause loss of efficiency and high VSWR.

The r-f impedance bridge is used in making these adjustments by connecting it across the input of the matching network with the transmission line disconnected. Where measurements indicate an input impedance of 70 ohms has been obtained (with antenna connected), the transmission line is reconnected for the final check before applying power from the transmitter.

Care should be taken that the capacitive branch uses capacitors of high enough current rating to hold up under the current at that point. Usually two or more capacitors of the proper value (to total the required value) are used to handle a larger current drain than one could safely carry.

Substitution Method. When no impedance bridge may be obtained or borrowed for the necessary length of time, a method commonly known as the substitution method is used to tune the antenna system. Fig. 20-35 shows the necessary arrangement of coupling circuit and tuning unit to carry out this procedure.

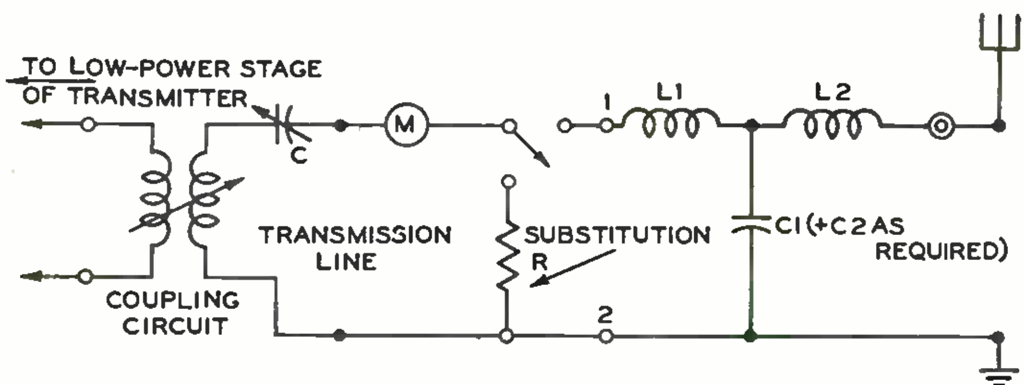


Fig. 20-35. Substitution method for tuning the antenna system.

It may be seen here that it is necessary to be able to switch the line from the tuner to a substitution *R* which should equal the line impedance, so that a change in line current as indicated by the meter *M* may be noted upon the switching action. The tuner is properly adjusted when no change occurs.

It should be obvious here that full power may not be applied to the line since the voltage rating of a nonreactive resistance is comparatively low. Therefore, connections must be made from a low-power stage in the transmitter.

Capacitor C is first tuned for maximum current reading in line meter M with the switch in the resistor position. The tuner input is then switched in, and C must be adjusted to again affect maximum line current. The change in the capacitance of C indicates whether the antenna circuit reactance is capacitive or inductive. If it is necessary to *increase* the capacitance, the load is capacitively reactive. Conversely, if the capacitor must be *decreased* to increase line current, the load is inductively reactive. If no change in C is necessary, the load is resistive.

Final Check of Antenna Match. Whichever method of tuning has been employed, a final check of match conditions should be made before applying full power to the circuits. The measuring equipment should be removed, and low-range thermal milliameters inserted at each end of the ungrounded conductor of the line. Sufficient power is then applied from a low-power stage of the transmitter so that the meters give an adequate reading. When the tuning adjustments are correct, these readings will agree within 15 per cent, showing a properly untuned feeding match between line and tower networks.

Tower Lighting

The FCC Standards of Good Engineering Practice set out a precise outline of required tower lighting as follows:

The following standard lamps and paints shall be used for the type of marking specified:

1. *Painting.* Each tower shall be painted throughout its height with alternate horizontal bands of international orange and white, terminating with orange bands at both top and bottom. (Orange yellow No. 5 color card, supplement to U.S. Army Quartermaster Corps Specifications No. 3-1.) The width of the orange bands shall be approximately one-seventh of the height of any structure less than 250 feet in height, and between 30 and 40 feet for structures over 250 feet in height. The width of the white bands shall be one half that of the orange bands.

2. *Lighting.* a. Towers, the over-all heights of which do not exceed 200 feet, shall be lighted as follows:

At the top of each tower there shall be installed two 100-watt lamps, enclosed in aviation red prismatic obstruction light globes. At least one of these lamps shall burn continuously from sunset to sunrise.

When only one lamp is operated, the circuit shall be equipped with a relay for instant automatic switchover to the other lamp in case of lamp failure.

At both the one-third and two-thirds levels of each tower there shall be installed two 100-watt lamps enclosed in aviation red prismatic obstruction light globes, at each level, one each on diagonally opposite corners of the structure.

All 100-watt lamps shall be type A-21 clear bulb traffic signal lamps (2,000 hours or equal).

All lamps shall be enclosed in aviation red prismatic obstruction light globes, and all lighting shall be exhibited from sunset to sunrise.

Special conditions of terrain and location with respect to airports or airways may require additional lighting of the character hereinafter specified for towers the over-all height of which exceeds 200 feet.

b. Towers, the over-all heights of which exceed 200 feet but do not exceed 300 feet, shall be lighted as follows:

At the top of each tower there shall be installed a hazard beacon similar and equal in effectiveness to the standard 300 m./m. airways electric lantern. This beacon shall flash and shall be equipped with two 500-watt lamps (both lamps to burn simultaneously) and aviation red color shades. The 300 m./m. electric code beacon shall be equipped with a flashing mechanism producing 40 flashes per minute, having a luminous period of 1 second, and a period of darkness of one-half second.

Towers over 300 feet in height (and less than 300 feet in height where special conditions obtain) may be required to install additional marking as follows:

General. Under particularly hazardous conditions and in areas of heavy air traffic, it may be necessary to add a 24-inch, 500- or 1,000-watt red rotating beacon equipped with an automatic lamp changer, to mark the installation. The beacon may be installed on the roof of the transmitter building, provided the point will afford proper visibility, otherwise it may be necessary to install the beacon on a separate tower of proper height or on the radio tower itself. The recommended setting of this rotating beacon is such that the center line of the light beam shall be approximately 3° above horizontal.

Adequate red warning lights shall be placed on the structure or structures during the period of construction and until the specified obstruction lights are installed and in operation.

Under certain conditions, it may be required to have the lighting controlled by a light-sensitive control device, the adjustment being such that lights will be turned on at a north sky light intensity level of 20-foot candles and turned off at a north sky light intensity level of 40-foot candles.

Under certain conditions, it may be required to install and operate a radio obstruction marker of a type to be specified.

Foregoing specifications require that at both the one-third and two-thirds levels of the structure there shall be installed two 100-watt lamps enclosed in aviation red prismatic obstruction light globes, at each level, one each on diagonally opposite corners of the structure.

Under certain conditions and in the case of triangular and other towers of unusual design, it may be required to install 100-watt lamps enclosed in aviation red prismatic obstruction light globes at various corners and levels throughout the height of the structure in such manner that adequate visibility and definition would be obtained from all angles of approach.

Towers of excessive height or peculiar (unusual) design will be made the subject of special investigation and analysis to determine proper marking and lighting.

In addition to the standard marking and lighting of the main structure, towers using guy wires will be made the subject of special study to determine proper marking and lighting of the guy wires.

The specifications relative to obstruction marking and lighting of radio towers of a certain height are general, and in locations of extreme hazard to air navigation, it may be required to install additional equipment on any height tower, in order to obtain suitable protection.

1. Type A-21, 100-watt traffic signal, clear, medium screw base lamps enclosed in aviation red prismatic obstruction light globes shall be employed in all cases where 100-watt lamps are specified.

2. Type PS-30, 200-watt general lighting service, clear, mogul prefocus base lamps shall be employed in all cases where code beacons with 200-watt lamps are specified.

3. Type PS-40, 500-watt general lighting service, clear, mogul prefocus base lamps shall be employed in all cases where code beacons with 500-watt lamps are specified.

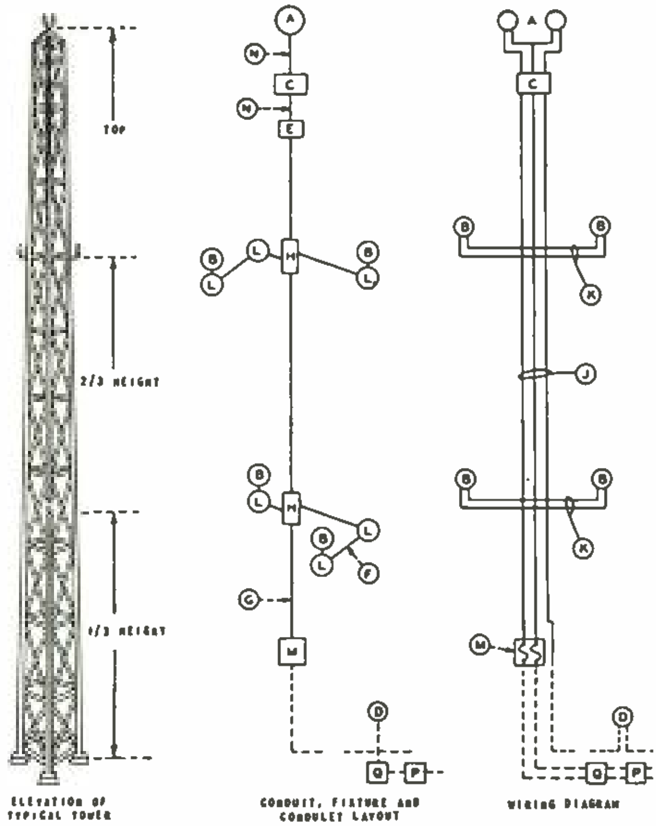
4. Type T-24, 500-watt aviation lighting service, clear, mogul bipost base lamps shall be employed in all cases where rotating beacons with 500-watt lamps are specified.

5. Type T-20, 1,000-watt aviation lighting service, clear, mogul

bipost base lamps shall be employed in all cases where rotating beacons with 1,000-watt lamps are specified.

6. The tower paint shall be kept in good condition and repainted as often as necessary to maintain this condition.

Methods of Tower Lighting. Fig. 20-36 shows a typical tower lighting plan for towers up to 150 feet high. The description of the items shown is as follows:



Courtesy Crouse-Hinds

Fig. 20-36. Typical tower lighting plan for towers up to 150 feet high.

Item A. Double obstruction light with 100-watt A-21 bulb (see "Standards" above). The wire feeder size should be based on no more than a 5% voltage variation from the rated voltage of the lamp.

Item B. Single obstruction light with 100-watt A-21 bulb, same feeder wire requirements as above.

Item C. Relay cabinet with transfer relay. (See Standards.) This relay switches to spare lamp on top double obstruction light in case of failure of regular lamp.

Item D. Pilot light with 6-watt lamp which indicates status (open or closed) of load contactor. This item is inside the transmitter building.

Item E. Junction box for double obstruction light atop tower.

Item F. Rigid galvanized conduit as required.

Item G. Rigid galvanized conduit as required.

Item H. Junction box for ⅔ height lighting bay.

Item J. Conductor, #12 cable (Underwriters type RW).

Item K. Same as J. (Recommended to meet requirements mentioned in Items A and B.)

Item L. Obround condulet with cover, rubber gasket, and nipple.

Item M. Weatherproof fuse box with 2-pole enclosed cartridge fuse cutout for 10-amp fuse.

Item N. Conduit nipple and supports.

Item P. Photoelectric control, contact capacity 4 amps.

Item Q. 2-pole load contactor.

All installations do not use photoelectric control of tower lights, although this method is to be recommended. Such control automatically regulates lighting when actual outdoor visibility warrants, regardless of time of day. Some installations use an electric clock motor control, set for proper "on" and "off" times which, of course, varies monthly. (Refer to Standards.) Photoelectric controls are adjusted to turn lights on at a north sky light intensity of approximately 35 foot-candles, and off at a north sky light intensity of about 58 foot-candles.

R-F Chokes for Lighting Systems. Broadcast installations where the tower itself forms the antenna and is insulated from ground (all a-m towers except shunt-fed types) must employ a means of preventing the radiation energy from following the tower lighting system into ground, thus detuning the tower. This is accomplished by lighting filters consisting of inductors and capacitors.

The filter must present high impedance to r.f. while presenting negligible impedance to the 60-cps a.c. Such a filter also serves to drain static charges on grounded a-c lines. The coil must be able to carry adequately the amperage of the lighting circuit.

Fig. 20-37(A) and (B) illustrates the usual lighting filter circuit. The choke coils should have an inductance of no less than 350 microhenries at 1,000 kc. The capacitors are 0.01 μf.

Fig. 20-38 is a photo of the Austin type tower lighting transformer, which is a transformer of ring-type windings using an air-gap between primary and secondary rings. These rings are oriented at 90° to each other as shown, adding negligible capacitance at the tower insulating zone. This transformer requires no housing, chokes, or filters. The

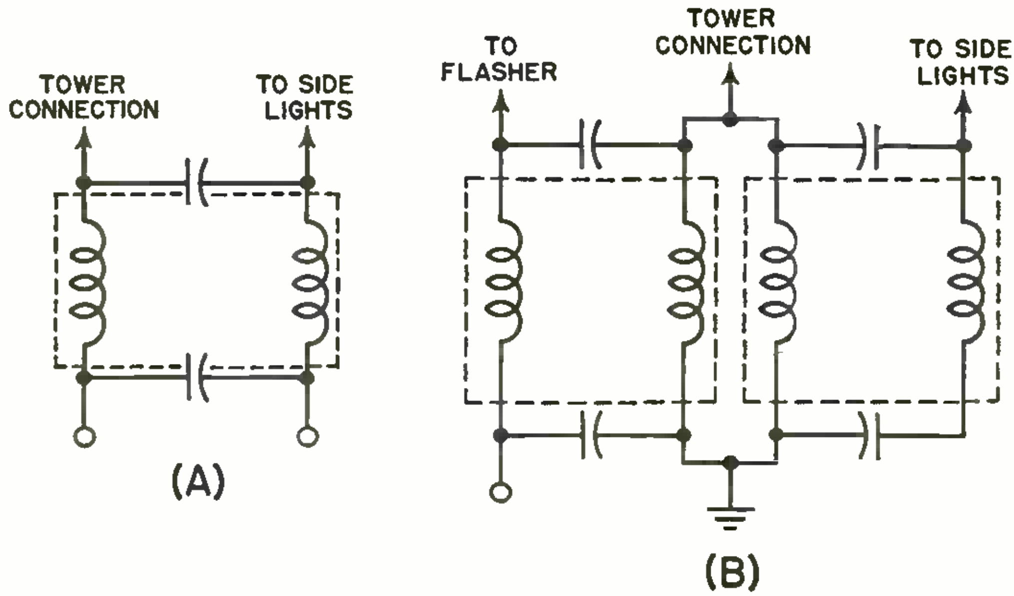


Fig. 20-37. Typical lighting filter circuits.

primary is usually attached to the base of the tower insulator or pier supporting the insulator. The secondary may be supported by a conduit attached to the top of the insulator or to the tower. The photo shows support by the conduit which then runs to the tower.

Fig. 20-38. The Austin type tower lighting transformer.

Setting Lightning Gaps. Gaps of either horn or ball types should be adjusted so that momentary 120% modulation peaks (positive peak reading) cause a spark to jump between the gaps. The final adjustment is then to move the gap "just a hair" farther apart. This method of adjustment will provide adequate protection from lightning surges under wide variations of temperature and humidity.

F-M ANTENNAS

F-m broadcast antennas operating in the 88 to 108 Mc region differ radically from the usual tower type of a-m operation. A high gain of power may be achieved by concentration of radiation in the horizontal plane at the expense of radiation in the vertical plane, which is considered useless for f-m broadcasting. The theory of the propagation of f-m power, since it is allocated to the vhf band, is based primarily on "line-of-sight" reception. Fig. 20-39 shows "optical path" calculating procedure and is self-explanatory. It should be remembered here that tall steel structures must be considered in practice when predicting field strengths of received signals. H_1, the height of the transmitter antenna, is from ground to the *effective radiating center* of the antenna elements.

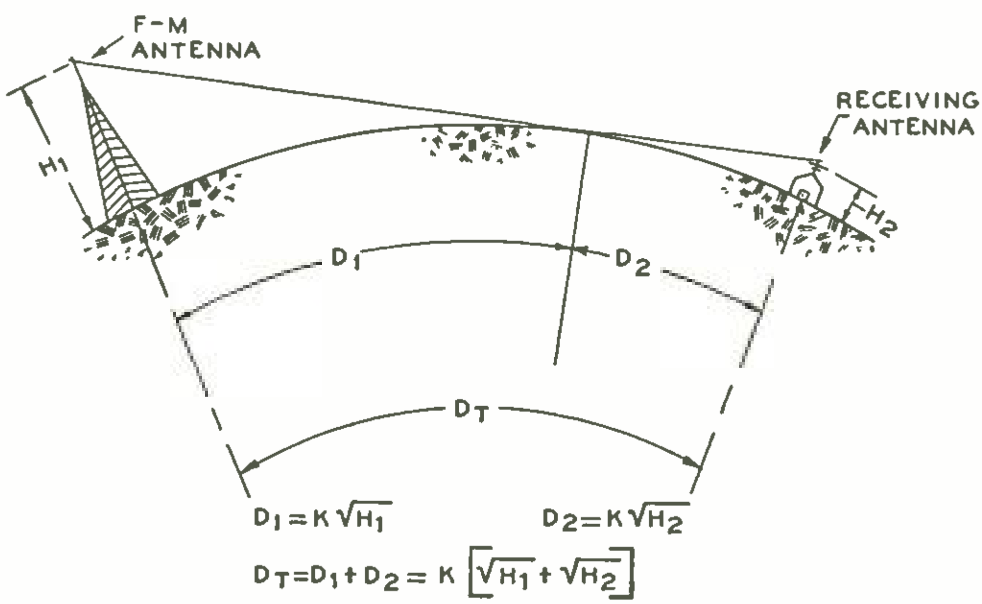


Fig. 20-39. "Optical path" calculating procedure for the propagation of f-m power.

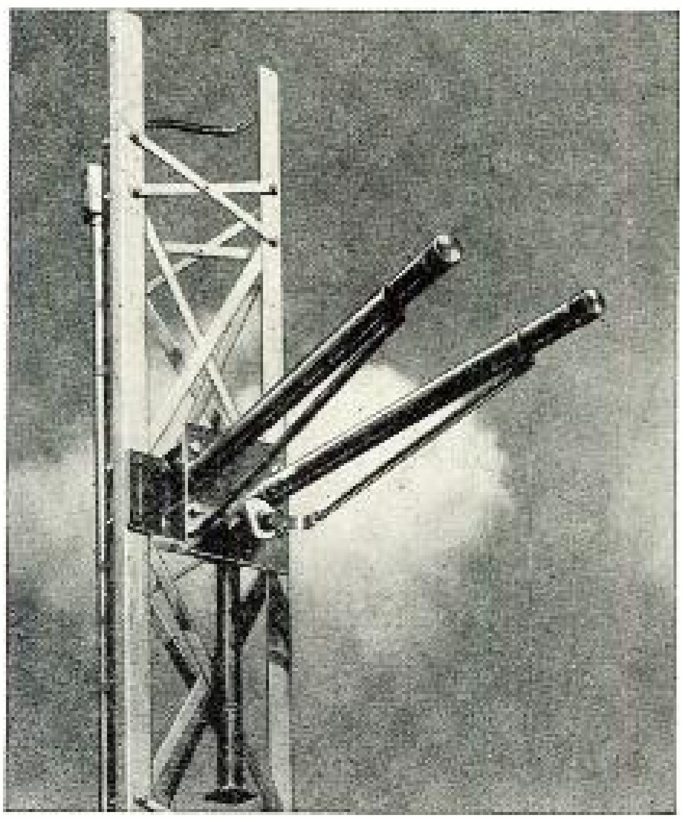
When high gain is used in the transmitter antenna, the usable range of the f-m signal is extended in practice over the theoretical calculation of Fig. 20-39. As a practical example, find the "line-of-sight" horizon for a 500-foot transmitting antenna and a 50-foot receiving antenna as follows:

$$D_T = 1.2\ [\sqrt{500}\ + \sqrt{50}\ ] = 1.2\ (29.3) = 35.16 \text{ miles.}$$

Therefore, the optical path is 35.16 miles. In practice, a 10-kw transmitter which was using an antenna with a power gain of 12 at 500-foot height, placed a 20 microvolt per meter signal at a distance of 88 miles. This is sufficient signal strength for most modern f-m receivers in rural or community locations. Predicting the range of f-m stations is discussed later.

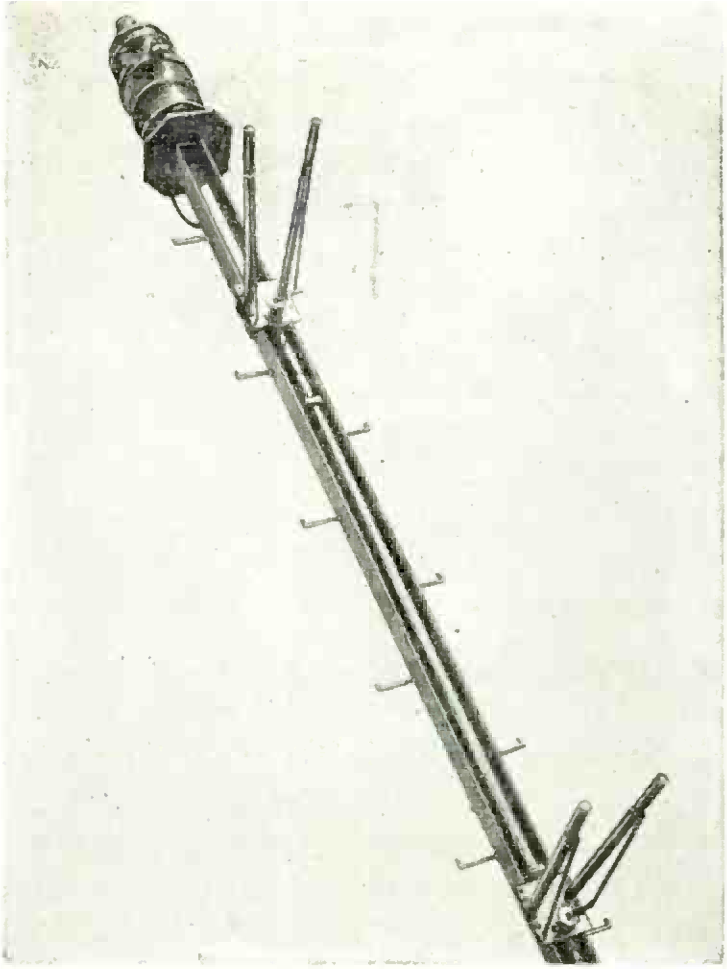
Types of F-M Transmitter Antennas

Andrews Multi-V. The Andrews Multi-V antenna is illustrated in the photos of Fig. 20-40(A) and (B). The radiating portion of the antenna elements form a V, operating on a folded dipole principle. Fig. 20-40(A) illustrates a single bay as mounted on the side of a tower. Fig. 20-40(B) is a two-bay antenna (this is the minimum number of bays used) as supported by an H beam atop a tower or other supporting structure. Fig. 20-40(C) illustrates the mechanical construction and simplified schematic circuit diagram.



Courtesy Andrew Corp.

Fig. 20-40(A). The V-shaped bay of the Andrews Multi-V antenna.

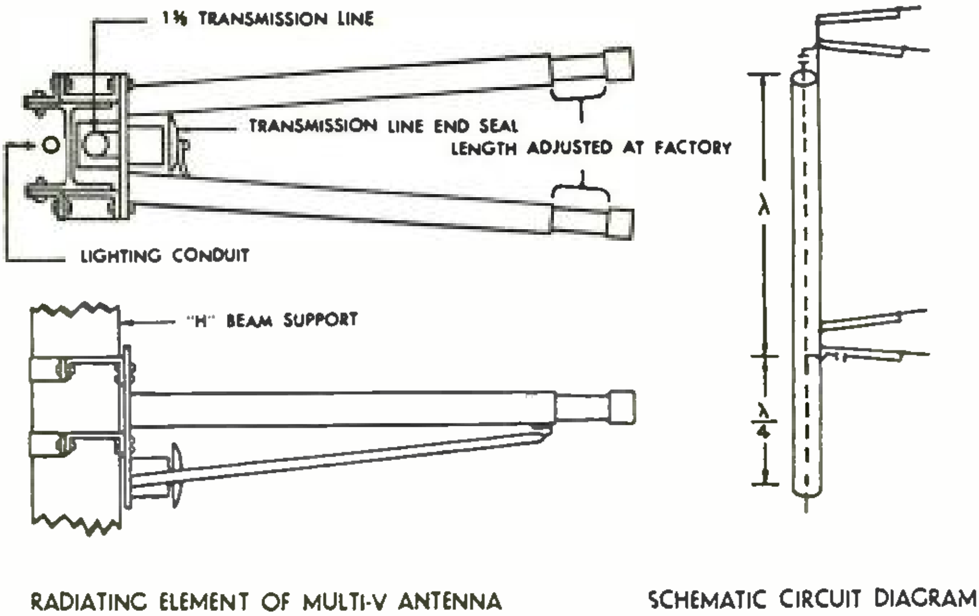


Courtesy Andrew Corp.

Fig. 20-40(B). The two-bay antenna mounted for f-m transmission.

The bays of the V antenna are connected and supported by a 1⅝ inch rigid coaxial line of 51.5 ohms impedance, being held in this fashion one wavelength apart. It is observed from the schematic of Fig. 20-40(C) that the bays are fed in parallel, with a series capacitor for each antenna element that forms one leg of the V. The one-quarter wavelength line below the first bay is a matching section that transforms the impedance at the junction point to 51.5 ohms, so that a standard 51.5-ohm line is used to feed the entire array.

In practice, only two different lengths of coaxial are used between bays (ideally one wavelength apart), to cover the entire f-m broadcast band. These lengths are 10.3 feet for 88 to 98 Mc, and 9.3 feet for 98 to 108 Mc. Tuning is accomplished by extending the secondary arms shown to a length dependent upon exact operating frequency. They are then soldered at the telescoping point to the primary arms to form a rigid and gas-tight connection.



Courtesy Andrew Corp.

Fig. 20-40(C). The bays are fed in parallel. A quarter-wavelength line matches impedance to a standard 51.5-ohm line.

To accommodate expansion, the lower bay is fastened rigidly to the tower, while the upper bay is held by a set of especially designed straps which permit vertical movement and prevent horizontal movement.

The power gain for a two-bay multi-V antenna is 1.6; for a four-bay 3.7; and for an eight-bay, a power gain of 7.3 is obtained. These are power gains referred to a dipole.

The RCA Pylon. The basic unit of the RCA Pylon f-m broadcast antenna is a steel cylinder approximately 13.5 feet high, with a narrow slot running from top to bottom. The theory of operation is based upon transmission-line theory and may best be explained by reference to Fig. 20-41.

Fig. 20-41(A) represents a simple loop of wire through which a current is passing, resulting in lines of force. The Pylon antenna itself may be represented by an infinite number of these loops stacked on top of each other, connected in parallel, approximately one wavelength high and one-half wavelength in circumference, as shown in Fig. 20-41(B).

As shown, the termination of the imaginary loops take the shape of a slot running the length of the cylinder. This slot may be considered an open wire transmission line which is fed at the center and shorted at both ends. The method of transmission-line connection at the center is shown in Fig. 20-41(C), and is located inside the metal cylinder. This results in the familiar voltage distribution curve of Fig. 20-41(D).

The currents which flow along the opposite edges of the slot are out of phase (one-half wavelength apart) and, therefore, cancel out as in any open-wire transmission line. The current which flows *around* the cylinder creates a long vertical shaft of magnetic lines of force, and since the electric field is always at right angles to the magnetic field, the radiation is horizontally polarized. Horizontal polarization is standard for f-m broadcast.

The above discussion then shows how vertical radiation is cut down, and the power applied in the horizontal plane which results in a power gain. The average power gains of the RCA Pylon are 1.5 for a single section; 3 for two sections; 6 for four sections. Eight sections (stacked on top of each other), results in a power gain of 12 times the applied power. Fig. 20-41 (E) shows the simplified schematic of a four-section Pylon, illustrating method of running the transmission line. Standard 51.5-ohm line of required power rating is used to feed the antenna.

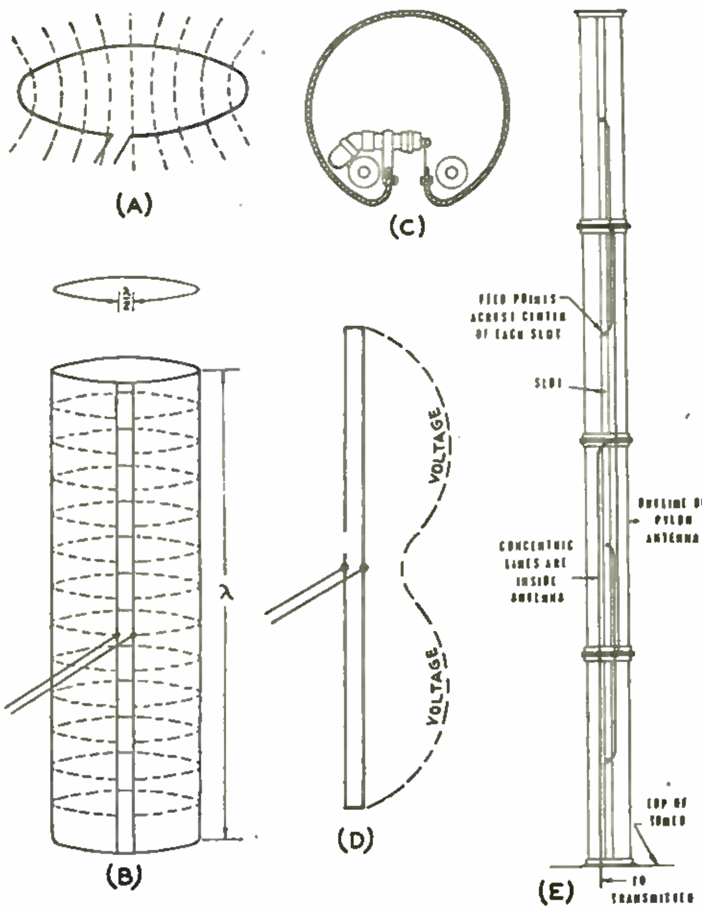
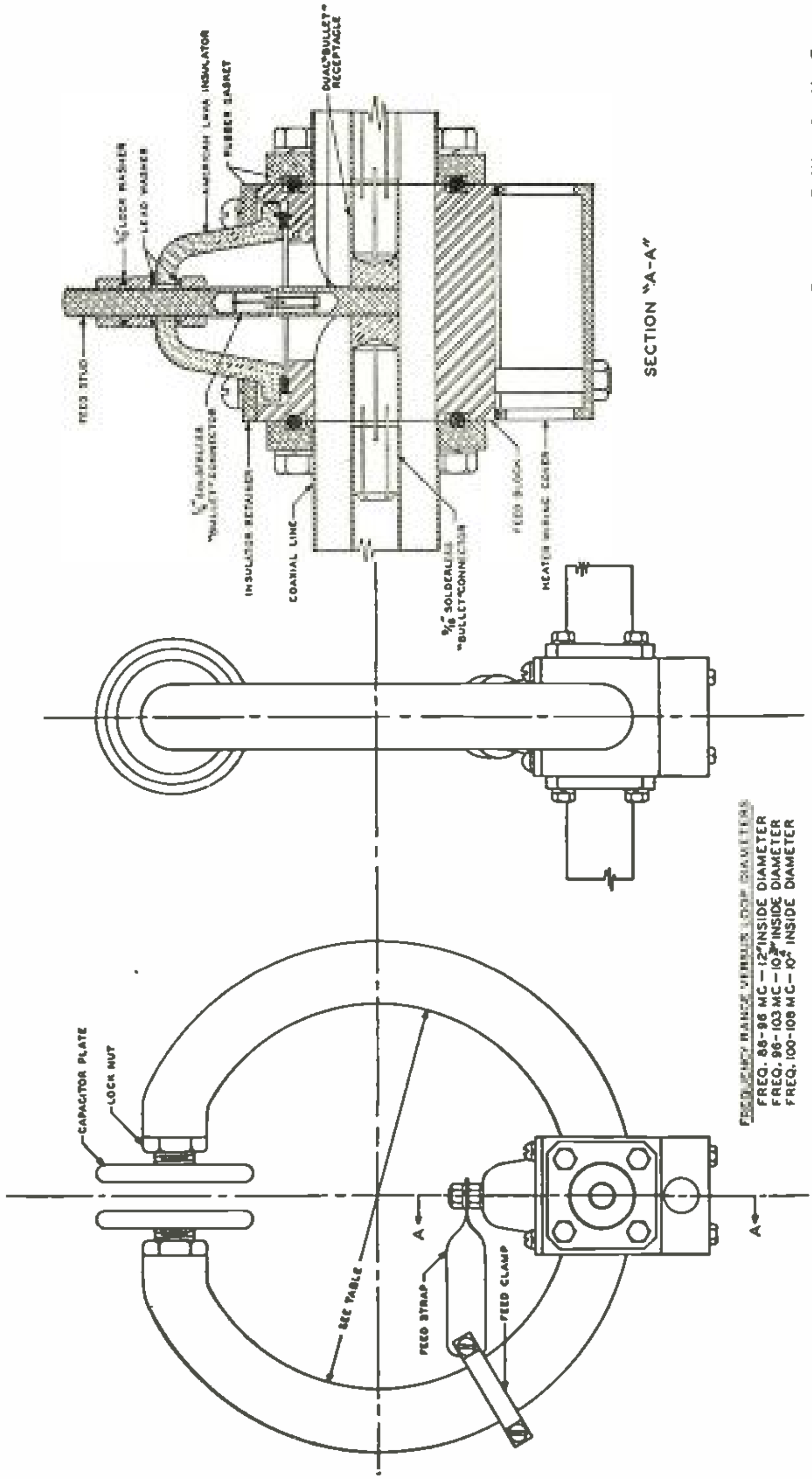


Fig. 20-41. The RCA Pylon f-m broadcast antenna.



Courtesy Collins Radio Co.

Fig. 20-42. The Collins 37 F-M Ring Antenna.

Collins 37 F-M Ring Antenna. This broadcast antenna consists of one or more radiating rings with a connecting interring transmission line, and may be mounted on the side or on top of a tower.

The radiating ring itself is of metal with a diameter depending upon frequency as shown in Fig. 20-42. A capacitor plate terminates each open end of the ring to provide exact tuning to the operating frequency. Electrically, the ring may be thought of as a folded dipole provided with concentrated end capacitance, bent to form a loop antenna, provided with a means of adjusting the end capacitor. Details of the mechanical construction are shown by Fig. 20-42.

The rings are mechanically supported by a single interconnecting feed line, spaced one wavelength apart and shunt fed. Standard 51.5-ohm line is used to feed the antenna system. Relative power gain per number of rings used is given in the Table 10.

Table 10

No. of Rings	Power Gain
1	0.9
2	2.0
3	3.0
4	4.1
5	5.1
6	6.2
7	7.2
8	8.3

F-M BROADCAST STATION DATA

Before taking up the problem of f-m station data as set forth by the FCC Standards of Good Engineering Practice Concerning F-M Broadcast Stations, the following definition of terms should be reviewed.

F-M Broadcast Station. A station employing frequency modulation in the f-m broadcast band (see definition below) and licensed primarily for the transmission of radiotelephone emissions intended to be received by the general public. This emission is classified F 3, as compared to A 3 for a-m stations.

Frequency Modulation. A system of modulation where the instantaneous radio frequency varies in proportion to the instantaneous amplitude of the modulating signal (amplitude of modulating signal to be measured after pre-emphasis), and the instantaneous radio frequency is independent of the *frequency* of the modulating signal.

F-M Broadcast Band. The band of frequencies extending from 88 to 108 Mc, which includes those assigned to noncommercial educational broadcasting. (See excerpts from F-M Rules and Regulations in the Appendix).

Center Frequency. a. The average frequency of the emitted wave when modulated by a sinusoidal signal. b. The frequency of the emitted wave without modulation.

Frequency Swing. Instantaneous departure of the frequency of the emitted wave from the center frequency, resulting from modulation.

F-M Broadcast Channel. A band of frequencies 200 kc wide and designated by its center frequency. Channels begin at 88.1 Mc and continue in successive steps of 200 kc to and including 107.9 Mc (See Appendix.)

Antenna Field Gain. Ratio of the effective free-space field intensity (see definition below) produced at one mile in the horizontal plane expressed in millivolts per meter for 1-kw antenna input power to 137.6 mv/m. The antenna field gain is the square root of the power gain.

Antenna Power Gain. The square of the antenna field gain. It is the ratio of the effective power at a given point to the power that would have been present using an ordinary dipole antenna.

Free-Space Field Intensity. The field intensity that would exist at a point in the absence of waves reflected from the earth or other reflecting objects.

Multiplex Transmission. The simultaneous transmission of two or more signals within a single channel. As applied to f-m broadcast stations, it means the transmission of facsimile or other signals in addition to the regular broadcast signals.

Percentage Modulation. Ratio of the actual frequency swing to the frequency swing defined as 100 per cent modulation, expressed in percentage. For f-m broadcast stations a frequency swing of ± 75 kc is defined as 100% modulation.

Effective Radiated Power. The product of the antenna power (transmitter output power less transmission-line loss) times (a) the antenna power gain or (b) antenna field gain squared.

Service Area. Service resulting from an assigned effective radiated power and antenna height above average terrain.

Antenna Height above Average Terrain. The average of the antenna heights above the terrain from 2 to 10 miles from the antenna. (In general, a different antenna height will be determined by each direction from the antenna. The average of these various heights is considered as the antenna height above average terrain.)

Estimating F-M Broadcast Station Coverage

Fig. 20-43 is used either to determine a given field-strength contour, or to find the required (effective) radiated power to produce a given field strength at a certain distance, for a given antenna height.

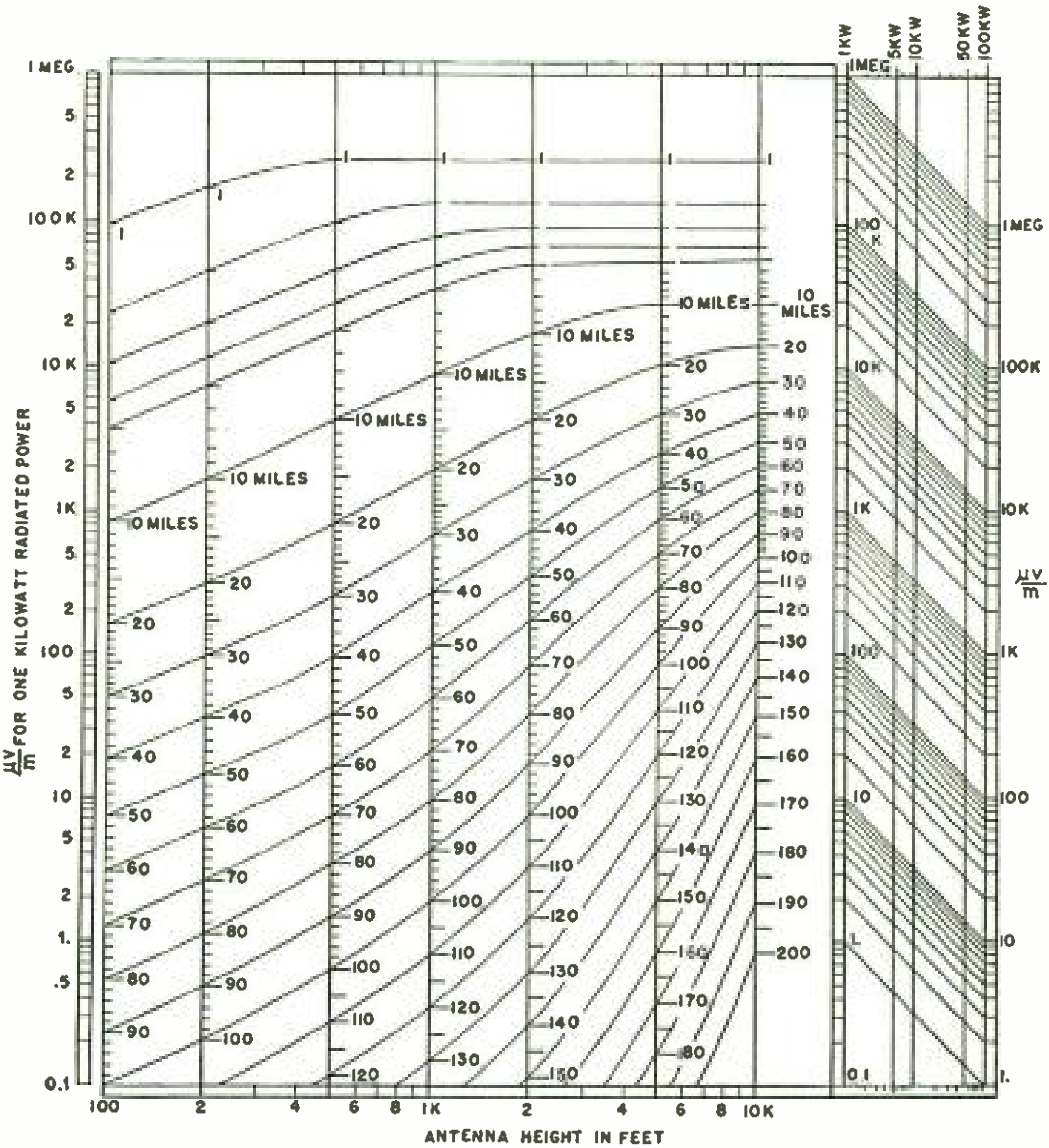


Fig. 20-43. Curves relating field-strength contour to radiated power and antenna height.

As an example of its use to estimate an approximate radius of an area within a given field-strength contour, consider the following problem:

1. To determine the distance from the antenna to the 1,000 mv/m contour when antenna height above average terrain is 500 feet, and effective radiated power is 1 kw. This point (distance) would probably vary along each radial unless land is very flat.

a. Place a straight edge along the horizontal plane (Fig. 20-43) from 1,000 on right edge to 1,000 on left edge. Mark the point at which the vertical line representing 500 feet (noted along bottom of graph) intersects the straight edge. It will be noted in this example that approximately 19 miles is the distance along this particular radial to the 1,000 mv/m contour. The graph is drawn on the assumption of 1-kw power, any other power requiring the use of the scale to the right on the graph. For example, the straight-edge run through 50-kw ordinate line and 1,000 mv/m line on extreme right, intersects the 500-foot height line at approximately 38 miles.

2. To find the value of the necessary effective radiated power necessary to produce a field intensity of 50 mv/m at 50 miles from an antenna 500 feet high.

b. Follow the 500-foot mark on the "antenna height" scale upward and locate the 50-mile point. Lay a straight edge (preferably transparent) across the chart from the point being sure it is parallel to the bottom line. Mark the point where it intersects with the diagonal line representing 50 mv/m and then from this point place the straight edge vertically on the chart and read approximately 2 kw required radiated power.

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INDEX