SECTION 3

SPECIALIZED BROADCAST RADIO ENGINEERING

AUDIO EQUIPMENT

307 D

- TABLE OF CONTENTS -

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TYPICAL STUDIO LAYOUT

GENERAL DESCRIPTION.--In previous technical assignments of this course the subjects of loudspeakers, microphones, amplifiers, and studios were discussed and their design and/or operation analyzed. These components, however, are only parts of the complete speech equipment, which comprises, in addition, connecting means and control equipment. It is the purpose of this technical assignment to study those components not previously discussed, and to see how the various parts fit together and function as an integrated whole.

It will be found that there is less uniformity in the studio layout than in the case of the transmitter. This is in part owing to the fact that the engineers of each station have different ideas as to the arrangement of the equipment, and in part owing to the fact that manufacturers in the past did not deem it advisable to build complete studio facilities for the smaller stations.

More recently these manufacturers have found it profitable to devote their time and energy to the development of more standardized equipment for the smaller stations, and some semblance of uniformity has appeared in the studio layouts. However, practice differs to some extent between the various manufacturers, and even in the case of the larger network chains, constant improvements in the broadcasting art have tended to make the later stations differ in some important details from the earlier stations of the network.

Essential Components of a Studio Layout.--Nevertheless it will be found that all speech equipment follows a more or less standard pattern, and that variations are mainly in the details, particularly in the control equipment. The basic components are shown in Fig. 1.

The microphones in the studio connect through their respective cables to wall or floor receptacles, from which run lines to separate pre-amplifiers located in the amplifier rack in the Control Room. Thus the output of each microphone is raised by its associated pre-amplifier to a higher level better suited for subsequent mixing, etc.

The outputs of the pre-amplifiers are then mixed or combined into one channel by means of a network known as a mixer. This is located on a console in the Control Room. The studio engineer sits in front of this console so as to be able alternatively to observe the action in the studio or the reading of his volume indicator, a meter--usually of the rectifier type--that indicates the output level of the signal en route to the transmitter.

He can also adjust the output of each microphone by means of an associated mixer or fader control located in the mixer, so as to achieve the desired tonal balance between the instruments or voices picked up by the respective microphones. In addition, he can adjust the combined output level ten necessary for the studio engineer or the program director to interrupt the program in order to advise the artists as to

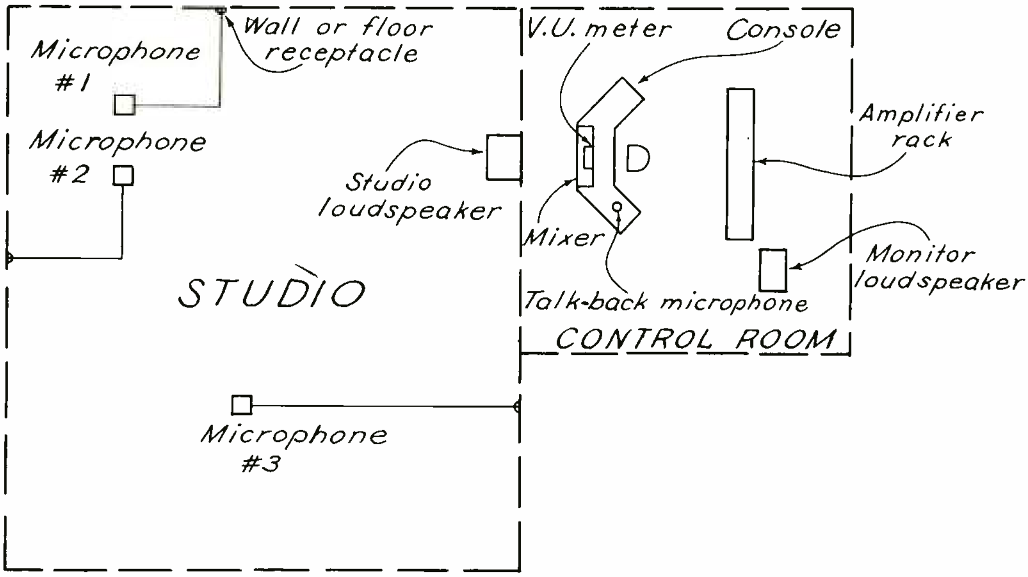


Fig. 1.--Physical layout of studio.

by an overall or Master Gain Control located in the mixer, and usually he "rides gain" on this control during the program presentation in conjunction with the V.I. (volume indicator).

To aid him in judging the quality of the sound, a high-quality monitor speaker is located in the Control Room. It is fed from a power amplifier located in the amplifier rack (or sometimes in the speaker cabinet). The power amplifier "bridges" across the outgoing audio line and thus also indicates whether or not the signal is proceeding to the transmitter.

During rehearsal it is often necessary for the studio engineer or the program director to interrupt the program in order to advise the artists as to some defect in their presentation. This is accomplished by the Talk-back Microphone in conjunction with a suitable switching arrangement, which connects the Studio Loudspeaker instead of the Monitor Loudspeaker to the power amplifier and the Talk-back Microphone instead of the studio microphones to the amplifying system. Note that it is necessary to disconnect the Monitor Loudspeaker when the Talk-back Microphone is operative in order to prevent acoustic feedback, since both units are in the Control Room.

The Talk-back Microphone can also be used to make an announcement from the Control Room (Control Room Announce) in an emergency, such as the sudden death or illness of the announcer in the studio. In some small stations the studio engineer also functions as the announcer, in which case he will make his announcements via the Talk-back Microphone.

In addition to the above there are various accessory pieces of equipment, such as special ON AIR and AUDITION studio lights to inform the artists whether they are on the air or merely on audition (rehearsal), control relays and push buttons to perform the various switching operations, and very often a transcription turntable console, that is located in the Control Room, and is operated by the studio engineer when a live talent program is not scheduled.

In many smaller stations a single Control Room may be placed between two studios, so that one or the other studio or both may be monitored. It is also usually possible to connect the Monitor Loudspeaker to another studio or an incoming live program in order to cue it. This means that the studio normally associated with the Control Room is not at the time under consideration on the air, but is awaiting the completion of a program originating at another point before doing so. Accordingly the studio engineer listens in or "cues" on his Monitor Loudspeaker on the present program originating elsewhere, and at the proper time signals his studio to start their program, after switching the transmitter line from the other studio or remote pickup to his studio.

Master Control Room.--In larger installations incorporating several studios, it is found advisable to relieve the individual studio engineer (or announcer) of the need for connecting or disconnecting his particular studio from the transmitter, so that he can devote his entire time to monitoring the program. In this case a separate room, known as the Master Control Room, houses most of the switching gear, and a separate crew of men, known as the Master Control Operators are employed to route the various programs. A possible setup is the following: three studios, A, B, and C, are to feed a network and the station or local transmitter. Assume that for the first half hour, studio A is to feed the network chain, and studio B is to feed the local transmitter, whereas for the second half hour studios A and B are to be off the air and studio C is to feed both the local transmitter and the network. Since the switching involves the concerted action of three studio engineers, it is preferable to have such switching performed by one man--the Master Control Operator--who will see to it that such switching is done with accuracy and dispatch. In addition, he will be able to hold up the program during the station "break", when all the individual stations momentarily disconnect from one another in order to announce their individual call letters, and see that all the stations are reconnected to the network before permitting the next program to start.

Another function of the Master Control Operator may be that of presetting the connections for the next program. This is true where the control circuits involve the preset feature. Thus, suppose many stations and programs are involved, such that during the first program time a certain number of stations are fed one program; a certain number, another program; a third group, another program, etc. Then, suppose at the end of the programs, the various stations are to be reconnected to the various program sources (studios and remote pickups) in another sequence. If the Master Control Operator were required to perform all the switching operations during the few seconds of the station break, he would have to work so fast as to be in danger of making errors in his switching operations.

To prevent this, an arrangement is made whereby during the new sequence, except for the fact that a master switch is in the position that permits only the first set of connections to function. (An actual circuit will be given farther on.) Thus the Master Control Operator can leisurely and carefully "punch up" and then check his new set of connections, and at the end of the program produce an almost instantaneous switchover by throwing the master switch. It is apparent that such preset facilities make for smoothness of program changing with a minimum of errors in switching.

BLOCK DIAGRAM OF AUDIO-FREQUENCY COMPONENTS.--In Fig. 2 is shown a block diagram of the audio-frequency components in the audio

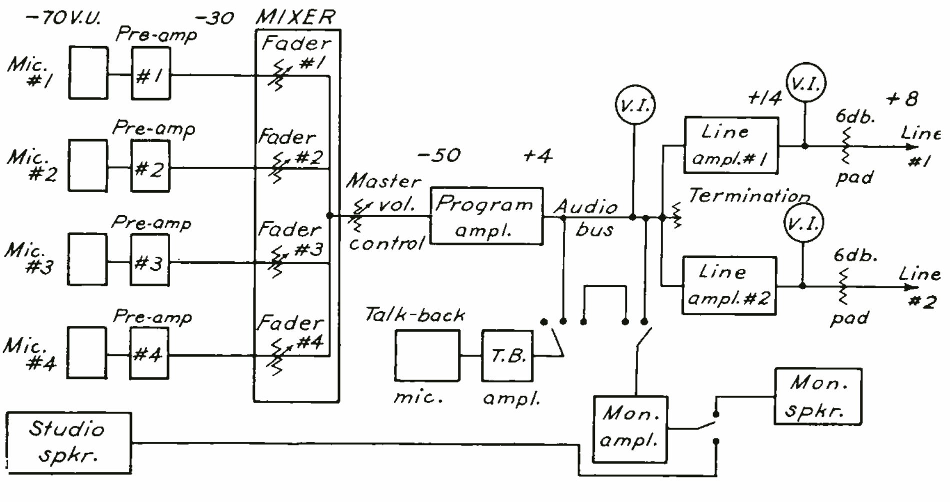


Fig. 2.--Electrical layout of studio speech components.

first program the Master Control Operator can operate a set of duplicate switches on a second switching panel, which would reconnect the stations and services in the chain. Four microphones or other sources of audio signal feed individual pre-amplifiers, whose outputs in turn are combined in the mixer through individual volume controls called faders into a single signal. The level of this is then controlled by a master volume control, and the output of the latter is then amplified by the Program or Studio Amplifier. Representative values of the levels in V.U. at the various points are indicated.

The output of the Program Amplifier is terminated in the proper value of resistance as shown. Across this output line, called an "Audio" or "Program Bus" are bridged several amplifiers. In Fig. 2 two Line or Channel Amplifiers and one Monitor Amplifier are shown. These bridging amplifiers are characterized by a high input impedance (as will be discussed later) and hence do not appreciably shunt the terminating resistor, so that it is immaterial to the Program Amplifier as to how many or how few of these amplifiers (within reason) are connected to the Audio Bus. The volume indicator (V.I.) also has a high impedance and can therefore bridge a line of nominal impedance—say 500 ohms—without appreciably affecting the level of the line.

Line Amplifier No. 1 may feed a network, to which a number of other stations are connected, while Line Amplifier No. 2 may feed the transmitter associated with these studios and located several miles away from the heart of the city. Each generally has a volume indicator connected to its output to check the level at that point with that at the output of the Program Amplifier. The meters are adjusted by means of external shunts so that even though there may be as much as 10 db difference in levels, they have the same scale deflection. Normal reading on the scale of 100 div. (about 3/4 full-scale) corresponds to +4 V.U. at the output of the Program Amplifier and +8 V.U. at the output of each line. (The V.I. meter has been described in Section II.)

It will be observed that an attentuation pad is interposed between each Line Amplifier and the line. The ordinary telephone line does not have a pure resistive input over the frequency range and hence is not a very satisfactory termination for the Line Amplifier. The latter, in turn, will not normally present the proper source impedance to the line if the line is made to appear twice the plate resistance of the output tube or higher. To offset such impedance mismatch, the 6 db pad is interposed. If a network has a sufficiently high attenuation, such as 6 db or more, then regardless of the value of the terminating impedance at one end, the impedance seen at the other end is approximately the characteristic impedance of the pad.

For example, the Line Amplifier may require a 500-ohm terminating impedance, and the telephone line may require a 500-ohm source impedance to obtain optimum results—such as minimum distortion and maximum flatness of frequency response—yet neither presents this value to the other. When a 6 db 500-ohm pad is interposed, however, each sees approximately 500 ohms looking back into the pad regardless of the actual value of impedance connected to the other end. The toll exacted is 6 db of attenuation, but this is not of any great consequence, since plenty of gain is available.

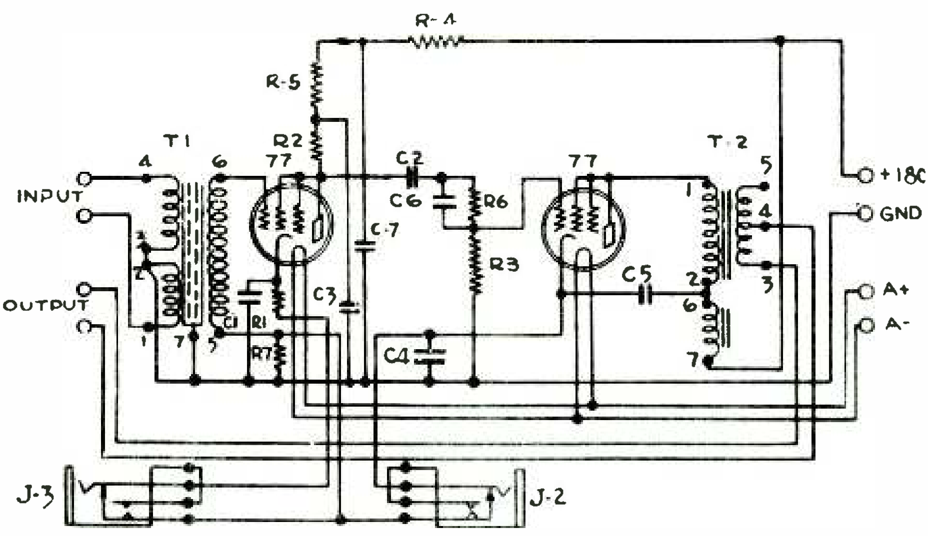
The Monitor Amplifier also normally bridges across the Audio

Bus, and normally feeds the Monitor Loudspeaker located in the Control Room. If, during auditions and rehearsals, the studio engineer wishes to speak to the studio, a switching circuit enables him to connect the Talk-back Microphone and Amplifier to the input of the Monitor Amplifier, and the output of the latter is at the same time connected to the Studio Loudspeaker instead of the Monitor Loudspeaker. For Control Room announce the Talk-back Microphone and Amplifier are connected to the audio bus and the control room speaker is disconnected to prevent acoustic feedback.

In somewhat more elaborate systems the Monitor Amplifier and Speaker can also be connected to an interpose a so-called Booster Amplifier between the fader controls and the Master Gain Control. This is necessary where the loss in the mixer is rather high. If the Booster Amplifier were not employed in such a case, the output of the Master Gain Control might be so low as to be dangerously near the inherent noise level in the system.

SPEECH EQUIPMENT

MICROPHONE PRE-AMPLIFIERS.-- Microphones have been discussed in a previous technical assignment. The associated microphone pre-amplifier is either a single-stage or two-stage unit and has a gain



(*Courtesy RCA*)

Fig. 3.—Type 41-B microphone pre-amplifier.

incoming line or another studio to cue a program originating at a point other than the associated studio. Another variation is to of about 40 db. A representative circuit is shown in Fig. 3. Note that the input transformer has no load resistor connected across the secondary. As was mentioned in the technical assignment on microphones, this gives a 3 db increase in signal-to-thermal noise ratio. It is designed to operate from a 44-A velocity microphone, and has an input impedance of 250 ohms. The output impedance can be either 250 or 500 ohms, depending upon whether terminals 3 and 4, or 3 and 5 are employed. The total gain is 42 db. The jacks permit the plate current of each tube to be measured by means of an external meter.

Note in particular C6 and R6. These act as a booster of the high frequencies, since C6 by-passes R6 more completely at the higher frequencies, and therefore permits more high-frequency voltage to appear across R3. A further point of interest is that the wiring follows a definite color code, so that where several microphones and pre-amplifiers are wired to a mixer, assurance is had that all microphones will feed any signal picked up in phase with one another. However, care must be exercised that the velocity microphones employed all have the same face toward the source. If any one microphone is rotated through 180°, it will develop a voltage 180° out of phase with the others. There will, of course, be a phase difference between the output voltages of the various microphones in any event owing to difference in path length from the source of sound to these microphones.

The amplifier can operate with a.c. on the heaters provided the heater transformer is located at least three feet away from the amplifier in order to avoid hum pickup by the latter, and the leads are shielded. Neither side of the heater circuit should be grounded, but instead a potentiometer should be connected across this circuit, and the arm connected to ground for hum adjustment. The plate supply can be obtained from a properly filtered rectified source.

MIXERS.--Mixers are for the purpose of combining the outputs of several microphones or other sources into one channel. In the past it was necessary to place the microphone close to the source of the sound in order to obtain a satisfactory signal-to-noise ratio. This was particularly true of the carbon button microphone, which generates a high noise (hiss) within itself. Another factor was that of obtaining a high ratio of direct-to-reflected sound, particularly in the case of the condenser microphone, which is practically non-directional at the lower frequencies.

Need for a Mixer.--Where the microphone had to be placed close to the source, several microphones were necessary in the case of a large source such as an orchestra in order to pick up the various instruments in the proper relative volume. However, no one microphone can be prevented from picking up some sound from another part of the orchestra, and--owing to path length difference--the resulting electrical output may be considerably out of phase with that of the other microphone assigned to that part of the orchestra. Partial cancellation can therefore occur, and this effect will vary with the wavelength, hence frequency of the sound.

It has therefore been deemed preferable to employ one microphone as far as possible, and thus there may seem to be no need for a mixer any more. This is not the case, however; numerous occasions arise where its use is highly desirable, if not absolutely necessary.

Consider, for example, an orchestra and soloist. Even though the entire orchestra can be picked up by one microphone, it is nevertheless usually desirable to employ a separate microphone for the soloist, in order to make him stand out more clearly against the background music of the orchestra. Another microphone can also be assigned to the announcer, and even if this is employed only when the other microphones are inactive, it is nevertheless desirable to have all microphones connected so as to avoid clicks that might otherwise occur if the microphones had to be switched on and off.

Mixers are also of great value in the stage presentation of large motion picture theatres, and in sound motion picture recording. For example, one or more microphones may be assigned to the orchestra in the pit or at one part of the stage, other microphones may pick up the players on the stage, still other microphones may pick up sounds from the wings, etc. It is thus clear that mixers are a very important component of any audio system today, and their design is of value to the broadcast engineer.

Mixing may be done directly at the microphone outputs, or at the outputs of their associated preamplifiers. The former is known as low-level mixing, and the latter as high-level mixing. High-level mixing is generally preferred because it affords a higher signal-to-noise ratio, particularly that component of noise developed by the volume controls. Low-level mixing, however, has the advantage that individual microphone preamplifiers are not required; instead, one booster amplifier following the mixer.

General Design Considerations.--Mixers may be of the electronic or of the resistance-network type. The latter are generally preferred for broadcast use because of their lower distortion content, greater flexibility as regards balanced- and unbalanced-to-ground circuits, number of microphones, and physical location, in addition to the fact that they are probably less delicate and more reliable than tubes, less microphonic, etc. Each type, however, will be analyzed.

The number of microphones or stations that are normally accommodated by a mixer for broadcast use is from two to eight, with four and six the most popular number. As will be seen farther on, a mixer must be designed for a maximum number of stations; if it is later desired to increase the number, extensive changes have to be made that constitute practically a redesign of the unit. Thus it might appear desirable initially to design the mixer for a large number of stations. The objection to this is that the greater the number of stations, the greater is the insertion loss of the mixer unit in the system, i.e., the greater is the inherent attenuation exacted by this device in the process of mixing the various signals while preserving impedance match at all points, etc. Hence a careful survey has to be made of the station as to its present and future needs not only in the initial choosing of the size and number of studios, but even in the determination of the maximum probable number of microphones to be employed in any one studio.

Parallel Mixer Design.—In order better to appreciate the design of a mixer, it will be well to choose a specific example. Suppose three pre-amplifiers are to feed a common output load, and it is decided to connect the three in parallel to the load, as indicated in Fig. 4(A). Let the impedance looking back into any of the three pre-amplifiers be Z_1. Although a mixer circuit could be built for three input impedances that were different, it is simpler to design one for which all stations have a common value of input impedance here designated as Z_1.

Another requirement that is usually specified is that the normal output levels of the various pre-amplifiers be the same. This insures an inherent balance between the various signals mixed. If microphone #2, connected to pre-amplifier #2, has for example a lower output level, then pre-amplifier #2 will be adjusted to have a compensatory greater gain so that its output will be comparable to those of the other pre-amplifiers.

With these facts in mind, consider what the impedance at terminals A-B is looking to the left. It is clearly the three pre-amplifier output impedances in parallel, or $Z_1/3$. If the output load is to match this value, its impedance Z_2 must also be $Z_1/3$. Next consider what impedance any pre-amplifier sees looking to the right into the mixer network. From the symmetry of the circuit it is clear that any pre-amplifier sees the same impedance as any other pre-amplifier.

In Fig. 4(A), pre-amplifier #1, for example, sees at terminals C-D the impedances of pre-amplifiers #2 and #3 in parallel with one another and with $Z_2 (= Z_1/3)$. This amounts to $Z_1/2$ and $Z_1/3$ in parallel, which is

$$Z_{CD} = \frac{(Z_1/2)\ (Z_1/3)}{Z_1/2 + Z_1/3} = \frac{Z_1}{5} \qquad (1)$$

Thus pre-amplifier #1, whose internal impedance is Z_1, sees an impedance of one-fifth this value

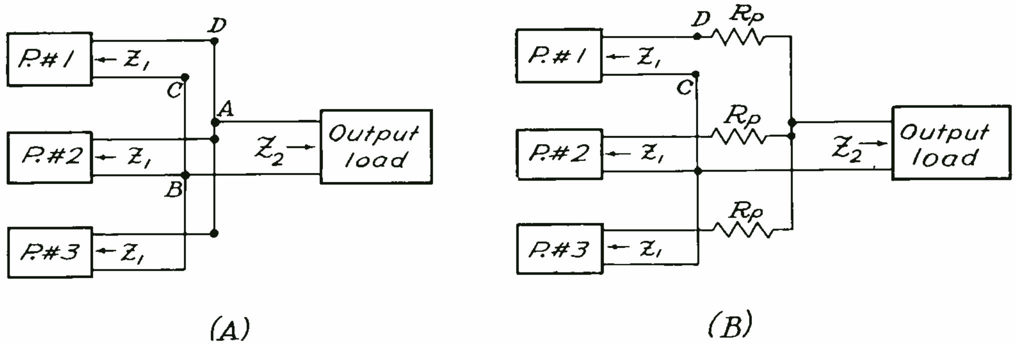


Fig. 4.—Derivation of parallel mixer circuit.

looking into the mixer, i.e., there is an impedance mismatch. Indeed, it is clear from the circuit of Fig. 4(A) that even if Z_2 were made infinite, the impedance seen by pre-amplifier #1 could not be greater than $Z_1/2$, that of the other two pre-amplifiers in parallel.

Such a mismatch is undesirable. To eliminate it, it is necessary to insert a certain value of resistance R_p in series with each pre-amplifier output as indicated in Fig. 4(B). If Z_1 is specified, then it will be found that Z_2 and R_p are determined thereby; or if Z_2 is specified, Z_1 and R_p are determined--in short, only one of the three impedances can be arbitrarily chosen, whereupon the other two are automatically fixed by such choice.

Normally Z_1 will be chosen to have some standard value, such as 250 ohms, or 500 ohms. Note that this refers to the impedance seen looking back into the output terminals of the output transformer of the pre-amplifier, i.e.,

$$Z_1 = r_p/n^2 \qquad (2)$$

where r_p is the plate resistance of the output tube ($2r_p$ for two tubes in push-pull) and n is the primary-to-secondary step-down turns ratio. This means that the output tube, when looking into terminals C-D Fig. 4(B), will also see an impedance Z_1, which will reflect to the tube terminals itself a value of r_p. Thus the tube will be terminated in a load resistance $R_L = r_p$ instead of $R_L = 2r_p$, as is normally specified for maximum undistorted power output. This is of no particular consequence, however, and facilitates the further design of the mixer. At the low levels at which such circuits operate, the tubes are in no danger of overloading, particularly where the output load (the mixer circuit) is a pure resistance. The impedance Z_2 is generally that seen looking into the primary of the input transformer of the Program Amplifier. This impedance is simply the reflected value of the load resistance connected across the secondary of the input transformer.

In terms of Z_1, R_p and Z_2 are found to have the following values:

$$R_p = Z_1\left(\frac{N_p - 1}{N_p}\right) \qquad (3)$$

$$Z_2 = Z_1\left(\frac{2N_p - 1}{N_p^2}\right) \qquad (4)$$

where N_p is the number of fader positions or stations. Examination of Eqs. (3) and (4) shows that as N_p increases, R_p approaches Z_1 in value, and Z_2 approaches zero.

Mention was previously made of insertion loss produced by the mixer. If only one pre-amplifier were employed to feed the load and the two were arranged to have a common or matched value of impedance, then maximum power transfer would occur and the insertion loss would be zero db.

An examination of Fig. 4(B), however, indicates that some of the power of any one of the pre-amplifiers is absorbed in the internal resistance of the other pre-amplifiers and in each R_p, so that only a fraction of the power output gets into Z_2. The logarithm of the ratio of the power getting into Z_2

as compared to that which would get to Z_2 if it equalled Z_1 and were alone connected to the pre-amplifier under consideration, is called the insertion loss. For the parallel type of mixer shown in Fig. 4(B), it has the value

$$\text{DB(insertion loss)}$$

$$= 10 \log(2N_p - 1) \qquad (5)$$

From Eq. (5) it is clear that as the number of stations (fader positions) N_p, is increased, the insertion loss is increased, and this justifies the statement made previously that a mixer should not be designed to handle more microphones than are deemed necessary in the studio.

Practical Parallel Mixers.--The circuit shown in Fig. 4(B) is but a simplified schematic of an actual parallel mixer. A more complete diagram is shown in Fig. 5 for a four-position mixer. It is practically a necessity that the volume of the individual pre-amplifier outputs be adjustable. This is accomplished by means of the individual fader variable tee pads F1 to F4. It will be recalled from Section II that such pads are made to provide attenuation in decibel steps between a source and a load of equal impedance. Under this condition the pad itself maintains such impedance match for all settings and merely varies the attenuation between the source and the load.

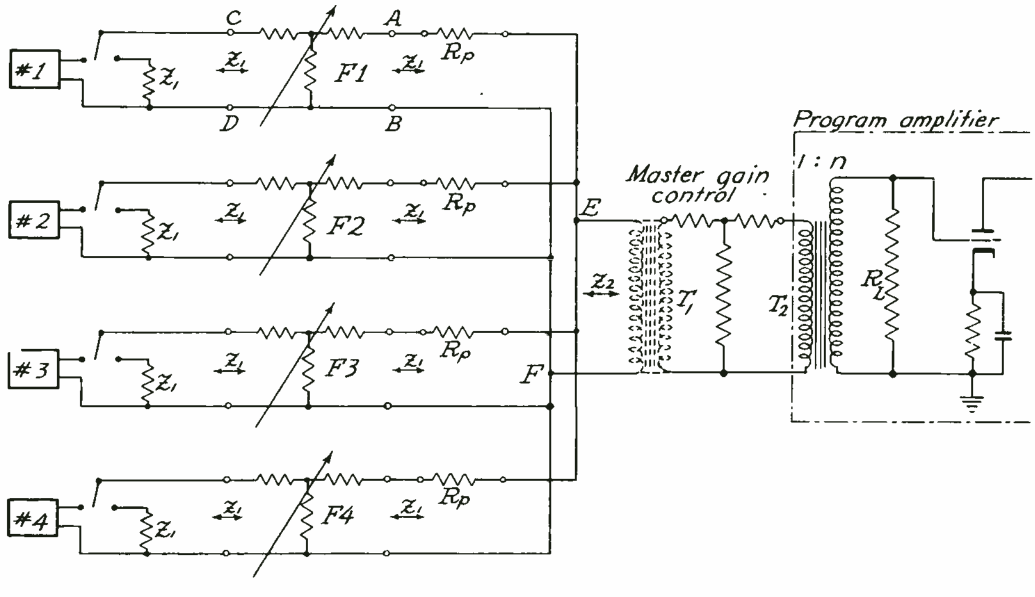


Fig. 5.--Practical parallel mixer.

If, however, the load or the source were mismatched to the specified characteristic impedance of the pad, then the impedance looking into either end of the pad would be other than its characteristic impedance, and moreover, would change with the pad setting. The significance of these remarks will be clear from the following discussion.

It has just been shown that by the inclusion of resistors designated as R_p, and by choosing these and the load resistance Z_2 as per Eqs. (3) and (4), both the sources (pre-amplifiers) and the load (input circuit of the Program Amplifier) will be matched by the mixer terminals to which they are connected. For example, the impedance looking into A-B to the right is Z_1 when R_p and Z_2 are properly choosen. Suppose Z_1 is taken as 500 ohms, i.e., the output impedance of each pre-amplifier is 500 ohms. Then the impedance looking into A-B to the right is 500 ohms. Now suppose tee pad F1 has a characteristic impedance of 500 ohms, i.e., it is designed to match 500 ohms at each end. Then if F1 is connected to A-B, the impedance looking to the right at CD will also be 500 ohms, and will therefore match the internal output impedance of pre-amplifier #1, which is 500 ohms. If F1 is varied so as to change the amplitude of the signal coming from pre-amplifier #1 into the mixer, the impedance looking into CD to the right remains at the value of 500 ohms.

An exactly similar process of reasoning applies to the impedance seen at A-B *looking to the left*. Thus, since 500 ohms are across CD (pre-amplifier #1), then the impedance looking to the left into F1 at terminals AB is also 500 ohms, and independent of the pad setting. Hence the mixer has a pre-amplifier and tee pad connected to its terminals AB that amount to a 500-ohm resistance. This being the case, it is immaterial to the other pre-amplifiers as to what the setting of F1 is, that is, varying the attenuation of F1 has no effect upon the signal levels of the other pre-amplifiers.

The same is true with respect to any other pre-amplifier: if the mixer is designed to match all impedances properly, then fader pads can be added to the basic circuit of Fig. 4(B) without upsetting the impedance matching by their presence or by their adjustment. Hence the impedance at CD is Z_1, looking either way, and the same is true at AB.

The same argument applies to terminals EF. The impedance seen looking either way is Z_2 ohms. Hence a tee pad of characteristic impedance Z_2 may be interposed here in order to vary the combined signal level; this is the Master Gain Control, and the variation of this control does not upset the impedances seen by the individual pre-amplifiers nor the level of the signal that they feed into the mixer. To summarize, by designing the mixer on a matched impedance basis, the adjustment of any one fader tee pad to a new value effects only the output of the associated pre-amplifier, and not that of any of the others, and the adjustment of the Master Gain Control affects the total output, but not the relative balance between the various pre-amplifier outputs.

For the four position mixer shown in Fig. 5, from Eqs. (3) and (4)

$$R_p = 500\left(\frac{4 - 1}{4}\right) = 375 \text{ ohms}$$

and

$$Z_2 = 500\left(\frac{2 \times 4 - 1}{4^2}\right)$$

$$= 218.7 \text{ ohms}$$

From Eq. (5) the insertion loss is

$$10 \log (2 \times 4 - 1) = 10 \log 7$$

$$= 8.45 \text{ db}$$

The value of R_p happens to be a standard value, 375 ohms. In another design it may come out an odd value, such as perhaps 286.7 ohms. While this can be rounded off to 287 ohms, or even 290 ohms without appreciably affecting the design, its non-standard value is of no great consequence since manufacturers at slight extra cost can build it to any value desired, or it can be built up from standard values, such as 200 + 75 + 15 = 290 ohms.

The fact that Z_2 may come out a non-standard value is a more serious matter. This is because attenuation pads and input transformers are generally built in standard impedance values, such as 50, 100, 125, 200, 250, 500 or possibly even 600 ohms. Any impedance other than a standard value involves additional costs, and also delay in ordering replacements so that an extensive stock may have to be carried.

In the example given above, Z_2 is 218.7 ohms. This is sufficiently close to 200 ohms, a standard value of impedance, to warrant the use of the latter value. However, it might be desired to use 500 ohms as the output impedance. In this case a matching transformer, T_1 in Fig. 5, would be employed, to change from 218.7 (or 200) ohms to 500 ohms. In that case a standard 500-ohm tee pad could be used for the Master Gain Control, and the input impedance of the Program Amplifier would also be this value.

In other examples Z_2 may come out some very odd value, such as 72.9 ohms, or perhaps 31.6 ohms. In such a case particularly the use of an impedance transformation to a standard value is advisable, and the proper matching transformer should be used. While it is true that this transformer (T_1 in Fig. 5) requires a non-standard primary winding, which increases its cost, nevertheless it permits the use of a standard impedance tee pad and standard input impedance for the Program Amplifier.

Observe that Z_2 comes out lower than Z_1, particularly for a large number of fader positions N. Also note that if Z_1 is chosen a standard value, such as 500 ohms in the above example, then Z_2 may come out a non-standard value, and necessitate the use of T_1 On the other hand were a standard value of impedance chosen for Z_2, then very likely Z_1 would come out a high and non-standard value, and necessitate individual matching transformers at the fader positions to permit the fader pads to be standard. It is therefore advisable to choose Z_1 and let Z_2 be determined by Eq. (4), since only one non-standard value may then be encountered.

In passing, it may be noted that a special parallel mixer design* is available for which Z_2 and Z_1 can both be arbitrarily chosen. The circuit, however, is limited to values of $N = 4$ or greater, and is then limited to ratios of Z_2 to Z_1 that are greater than certain minimum values. In addition, the insertion loss is greater than for the type of parallel mixer given above.

Furthermore, note the resistor marked Z_1 (Fig. 5) and the single-pole-double-throw switch at each fader position. These enable the microphone pre-amplifier to be disconnected from the mixer in order that it may be repaired, etc. By throwing the S.P.D.T. switch in the other direction, the fixed resistor Z_1 replaces that of the pre-amplifier, so that the mixer is still properly terminated, and operates so far as the other fader positions are concerned exactly as before. In many actual designs the fader pad is so designed that when it is turned down to zero transmission (infinite attenuation) further rotation actuates a S.P.D.T. switch mechanically incorporated in the pad (like an "on-off" switch on the volume control of a radio receiver) and performs the above switching operation. A further refinement is to have a second switch blade simultaneously short out the pre-amplifier output terminals and ground them so as more definitely to insure that no signal will then get into the mixer from that particular microphone owing to stray coupling. This helps avoid so-called "cross talk" in the mixer.

Finally, note that the actual termination is the reflected value of R_L, the resistor loading the secondary of the Program Amplifier's input transformer. Suppose T_1 were not used, and that the Master Gain Control is 218.7 ohms, or say 200 ohms ($= Z_2$). For the pad to present this impedance at its left-hand terminals to the mixer, it must be terminated at its right-hand terminals in 200 ohms. (This is not so important when the control is set for more than 6 db attenuation or thereabouts, but even then an impedance mismatch at its right-hand terminals, while it may not affect the impedance seen at the other end, will change the amount of attenuation produced at each step from the designed value.) The pad is therefore matched by using a turns ratio n for T_2 such that

$$n = \sqrt{R_L / 500}$$

If T_1 is also employed so as to transform Z_2 = 200 or 218.7 ohms to, say 500 ohms, then n must equal $\sqrt{R_L/500}$

Series Mixer.--Another form of mixer circuit is shown in Fig. 6. This is known as the Series Mixer, because the pre-amplifiers and associated fader controls are all connected in series to the output load Z_2. Four positions are shown, although no more than two are in general advisable in order that there be no cross talk, particularly from the end faders. This will be discussed farther on. Note in particular that the individual fader positions are shown as tee pads, and are connected in opposite fashion for the top and bottom

*See "Parallel Mixer Circuit Calculations"--C. W. Slaybaugh, *Broadcast News*, March 1940.

pairs. Also note that the Master Gain Control is an H-pad, and that the center tap of Z_2 (usually the primary of the input transformer of the Program Amplifier) is grounded. This, too, will be discussed more fully later.

In order to effect an impedance match at all points, *shunt* compensating resistors R_S are provided as shown. If Z_1 is chosen, then R_S and Z_2 are determined by the following equations

$$R_S = Z_1\left(\frac{N_S}{N_S - 1}\right) \qquad (6)$$

$$Z_2 = Z_1\left(\frac{N_S^2}{2N_S - 1}\right) \qquad (7)$$

where N_S is the number of stations or positions. Eqs. (6) and (7) involve N_S in reciprocal fashion as compared to Eqs. (3) and (4) for the parallel mixer. Accordingly, the greater the number of stations in the series mixer, the lower will R_S be, and the higher will be Z_2. In the case of $N_S = 4$, as in Fig. 6, and for $Z_1 = 250$ ohms,

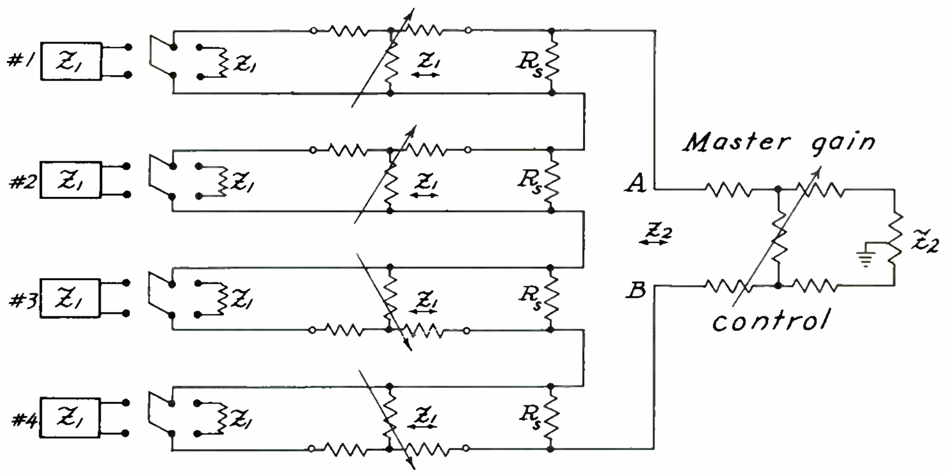


Fig. 6.--Series Mixer.

$$R_S = 250\left(\frac{4}{4 - 1}\right) = 333 \text{ ohms}$$

$$Z_2 = 250\left(\frac{4^2}{8 - 1}\right) = 571 \text{ ohms}$$

The odd value for Z_2 suggests that an impedance-matching transformer may profitably be interposed between terminals A-B and the Master Gain Control, so as to transform 571 ohms to a standard value, possibly 250 ohms, the same as the fader pads. A further refinement is to ground the *center-tap* of the primary of the transformer, and *one side* of the secondary, thus enabling a tee instead of an H-pad to be employed for the Master Gain Control. The suggested changes are indicated in Fig. 7. A transformer that permits such change in grounds from

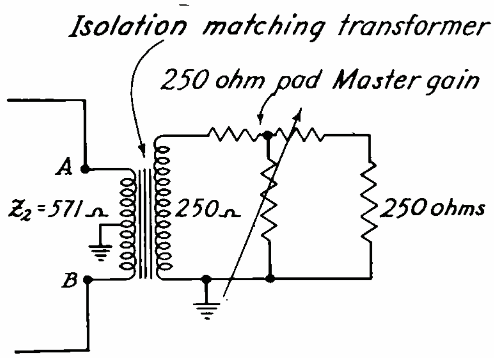


Fig. 7.--Modification in series mixer design.

one winding to the other is known as an Isolation Transformer.

The insertion loss of a series mixer is exactly the same as that of a parallel mixer, and is given by Eq. (5a), namely

$$\text{DB (Insertion Loss)} = 10 \log (2N_S - 1) \tag{5a}$$

For the four-position series mixer this will therefore also be

$$10 \log (8 - 1) = 8.45 \text{ db}$$

Series-Parallel Mixer.--The output of a mixer may be regarded as a new source impedance, and two or more mixers may therefore be mixed in an overall mixer. An elaborate system of this kind is employed in the Radio City Music Hall in New York City. Each mixer mixes the outputs of a group of microphones, such as those for the orchestra, or for the stage performers. An overall mixer then mixes these outputs and adjusts their relative volume (balance) for the sound reinforcement system employed in conjunction with the stage presentation. The output impedance Z_2 of each individual mixer is transformed to a common value, which becomes the Z_1 of the overall mixer, and for the latter, N represents the number of mixers that are to be mixed.

A similar line of reasoning permits a modification in mixer design known as the Series-Parallel Mixer. Suppose four positions are involved, i.e., two pairs of stations. Connect the individual stations in each pair in series, and then connect the two pairs in parallel. This is shown in Fig. 8.

Assume, for example, that the output impedance of each pre-amplifier is 500 ohms. From Eqs. (6) and (7), for $N_S = 2$, $R_S = 500(2/1) = 1000$ ohms, and $Z_2 = 500(4/3) = 667$ ohms.

This value of Z_2 becomes the new value of Z_1' for the parallel combination, for which Eqs. (3) and (4) are to be employed, and N_p is again equal to 2. From this $R_p = 667(1/2) = 333$ ohms, and $Z_2' = 667(3/4) = 500$ ohms. An important thing to note is that the output impedance Z_2' is exactly the same value as the input impedance Z_1, namely, 500 ohms. This will always occur if the total number of stations is a perfect square, like 4 above, or 9, 16, etc.

This type of mixer is used to quite an extent, although it has a higher insertion loss than the previous types of mixers, or a bridge type of mixer to be described next. The loss of this mixer is the sum of the losses of the two types of connections involved in its design, or

DB (Insertion Loss)

$$= 10 \log(2N_S-1) + 10 \log(2N_P-1) \quad (8)$$

For $N_S = N_P = 2$, Eq. (7) gives the value $2 \times 10 \log 3 = 9.54$ db which is greater than the previous value of 8.45 db for a four position series or parallel mixer.

the *parallel-series* instead of the series-parallel arrangement. For example, if $N = 6$, choose $N_P = 3$ and $N_S = 2$ rather than $N_S = 3$ and $N_P = 2$. Note that N must be an even number for either type of circuit.

For $N = 4$ the two types are equivalent. Farther on in this assignment, after all the different types have been discussed, a summary will be given in which the best circuit for a particular set

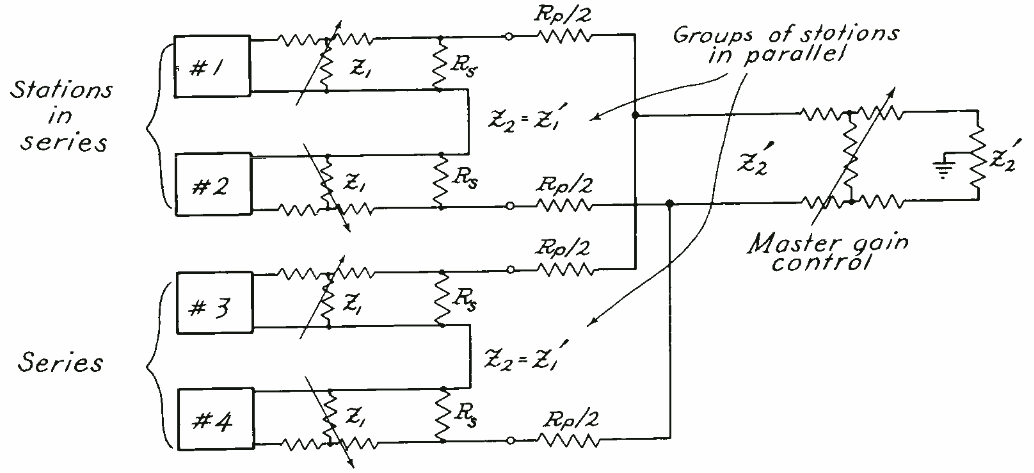


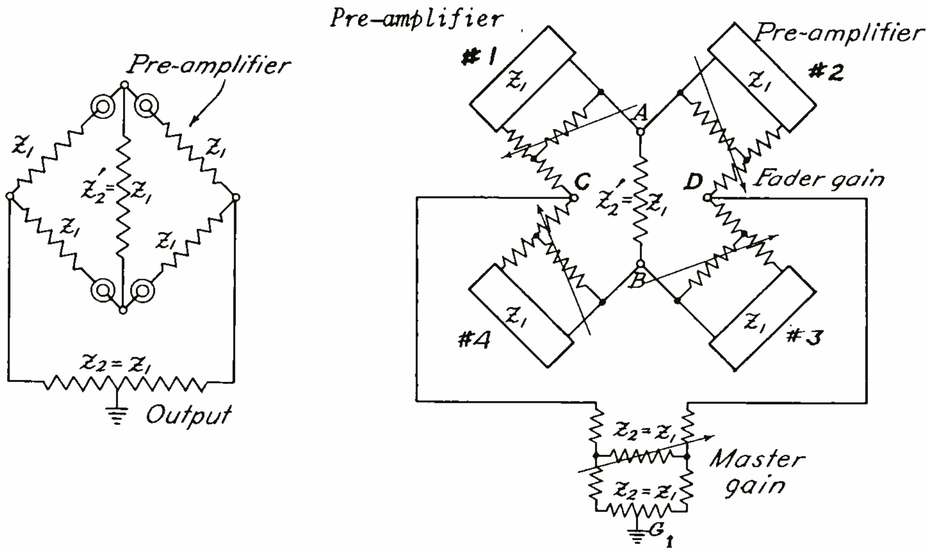
Fig. 8.—Series-parallel mixer.

One can also start with the stations in each pair connected in parallel, and then the two pairs in series. The values of R_P and R_S will be different, but the end value of Z_2' for a given value of Z_1 will come out the same if the overall N is a perfect square. Since the number of stations in series should preferably not exceed two ($N_S = 2$) in order to avoid cross talk, whereas the number of stations in parallel, N_P, can be any number, the better arrangement if N is greater than 4 is to use

of requirements will be indicated. It will then be found that neither the series-parallel nor parallel-series type of circuit is as desirable as other forms. However, these two types will be found in many actual installations, and hence the design factors are given here to acquaint the student with their calculations.

Bridge Type Mixer.--Another type of mixer fundamentally adapted for four positions only is shown in Fig. 9(A) and (B). It will be observed from Fig. 9(A) that the elements are in a bridge arrangement, with the pre-amplifiers acting as the arms. There are two outputs, one labelled Z_2 and the other Z_2'. For proper balance, these should have the same impedance as that of the pre-amplifiers, i.e., $Z_2 = Z_1$.

their normally grounded ends connect to terminal A or B. This is because from the symmetry of the bridge circuit, there will be no difference in potential between ground (terminal G_1) and either A or B; i.e., the latter are essentially at ground potential. However, neither A nor B should be directly grounded, for then elements of the bridge will be partially shorted out.



(A) Basic Circuit (B) Basic Circuit with Volume Controls

Fig. 9.—**Bridge type mixer.**

One of these (labelled "Output" in Fig. 9(A)), grounded at the center tap. No other direct grounds should be employed, so that if the vertical member marked $Z_2' = Z_1$ is also desired as an output, it must not be grounded at any point. Ordinarily this impedance would be a simple resistor, and only the horizontal member would be employed as an output.

In Fig. 9(B) is shown the same circuit with the fader and Master Gain Controls added. Note that the Master Gain is an H-pad, whereas the fader controls are tee pads, and that these are connected so that

One advantage of this circuit is that the output impedance Z_2 equals the input impedance Z_1, so *that all can be chosen of a standard value.* (However, the master control must be an H-pad, which is inherently more complicated and expensive than a tee pad.)

Another advantage is that the insertion loss is only 6 db, or less than that of any other type of four-position mixer. At the same time it is as free from cross talk as any other design (to be discussed later). Since four positions are a common value for a mixer, this circuit is of value in studio design. In Fig. 10 is shown ter Gain Control on the secondary side of the circuit. Moreover, the isolation transformer can be of standard impedance.

If an isolation transformer

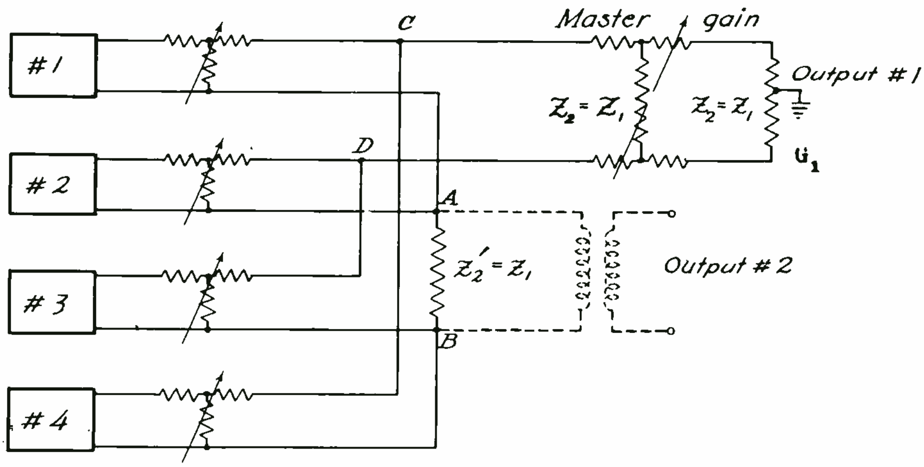


Fig. 10.—Usual schematic of bridge type mixer.

the more usual schematic arrangement for this circuit. Terminals marked A-B and C and D coincide with those similarly marked in Fig. 9(B) and will serve as a guide in identifying similar members.

Mention was made that the four pre-amplifiers can feed two outputs, but that only one can be grounded (Output #1 in Fig. 10). However, if an isolation transformer is employed as shown by the dotted lines, a second output can be obtained (Output #2), whose *secondary winding* can be grounded as desired; i.e., at the center tap, or one side. A similar transformer can be employed for Output #1, and the secondary can be grounded at one end, in which case a tee rather than an H-pad can be employed as a Master Gain Control

is used so that the secondary need not be grounded at the center tap, then four of the above mixers can become the input elements of a 16-position overall bridge type mixer, whose insertion loss is only 12 db, or less than any other type. Of course this principle can be carried on indefinitely.

Modified Series-Parallel Type.—A modified type of series-parallel mixer, mentioned by M. Rettinger,* is shown in Fig. 11. This circuit is characterized by an insertion loss that is lower than any other type except the bridge type, but

*See "Microphone Mixers", *Journal of the Society of Motion Picture Engineers*, Vol. XXVIII, No. 6, 1937, pp. 604-613.

unfortunately it is limited to an even number of positions.

The value of the compensating resistor R_p and of the output load impedance are

$$R_p = \frac{N - 3}{N} Z_1 \qquad (9)$$

$$Z_2 = \frac{4(2N - 3)}{N^2} Z_1 \qquad (10)$$

DB (Insertion Loss)

$$= 10 \log (2N - 3) \qquad (11)$$

where again N is the number of stations.

and

DB (Insertion Loss)

$$= 10 \log (12-3) = 10 \log 9$$

$$= 9.54 \text{ db}$$

Note that *the fader volume controls must be tee pads,* (unbalanced-to-ground inputs) arranged as shown, while *the master gain must be an H-pad* unless an isolation transformer is employed. In the above example Z_2 comes out equal to Z_1, a standard value, and this will also be the case for

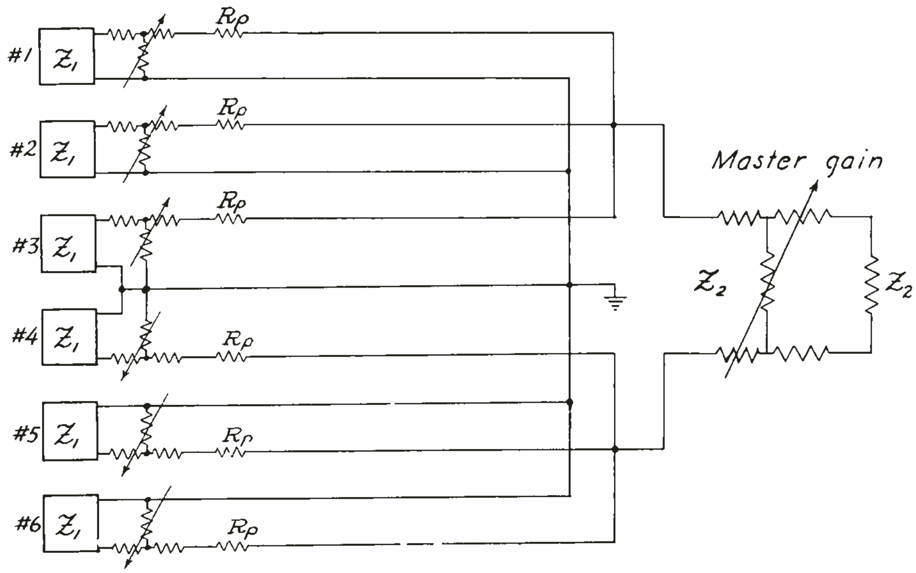


Fig. 11.—Modified series-parallel type.

For example, if a six-position mixer is desired, and $Z_1 = 500$ ohms, then

$$R_p = \frac{6 - 3}{6} \times 500 = 250 \text{ ohms}$$

$$Z_2 = \frac{4(12 - 3)}{36} \times 500 = 500 \text{ ohms}$$

$N = 2$. But for other values of N this will not in general be so. It will be found that the insertion loss of the modified series-parallel mixer is lower than that of a parallel mixer for the *same number* of stations, as is clear from a comparison of Eqs. (11) and (5). However, a check of these equations will also show that the insertion loss of the modified series-parallel mixer is equal to that of the parallel mixer having *one less station.* This means that if an odd number of stations *that are unbalanced-to-ground, is desired*, such as five, it is immaterial as far as insertion loss is concerned whether a parallel mixer of exactly five stations, or a modified series-parallel mixer of six stations is employed.

From a practical viewpoint the latter would be preferable. If only five microphones are actually to be used, a resistor can be used in place of a sixth microphone. Should it ever be desired to add a sixth microphone, the circuit is available without any redesign except the mere removal of the above resistor. Note, however, that the modified series-parallel mixer can be used only with unbalanced-to-ground inputs, as mentioned above; except, of course, if isolation transformers are employed.

Cross Talk.—Even if a mixer is designed according to the formulas given for any of the types discussed, it may still be unsatisfactory because of a factor known as cross talk. This is not peculiar to mixers alone, but may occur anywhere in the audio system; however it is often most noticeable in the mixer circuit. Cross talk refers to the pickup of unwanted signal from one channel in another channel. In a mixer, for example, one fader position may be turned off completely (fader pad set for infinite attenuation) and yet signals may appear in the output of the mixer from the supposedly dead microphone.

The cause appears to be mainly capacitive coupling between the preamplifier output cables running to the fader volume controls, and is due to improper design as regards types of pads used, method of connection in the mixer, and point at which the circuit is grounded. Accordingly a simple analysis will be given here as a guide in locating such faults in an existing or proposed mixer circuit.

In the technical assignment on microphones it was shown how a power circuit could produce unwanted noise in a microphone circuit owing to capacitive or inductive coupling, depending upon the impedance of the circuit. The same kind of coupling may give rise to cross talk (a special kind of noise pickup) between two audio circuits.

The reason that cross talk is of particular importance in the case of a mixer is that any fader pad may be set at infinite attenuation, to "kill" some particular microphone, and yet owing to coupling to another fader position, appreciable amounts of signal may enter the mixer, causing interference and possibly even embarrassment in the presentation of the program. Another factor is that for large amounts of attenuation in the fader pad, the unwanted transmission path may produce signals comparable to those passing through the pad, and since the unwanted path represents coupling that is more effective at the higher frequencies, a peaked rather than a flat high-frequency response may result.

Balanced and Unbalanced Circuits.—In order to appreciate cross talk phenomena, a brief review of balanced-and unbalanced-to-ground circuits is desirable. A circuit is considered balanced to ground when corresponding points of the two sides of the circuit have equal impedance and voltage to ground. Fig. 12 illustrates a typical balanced-to-ground circuit. Both the

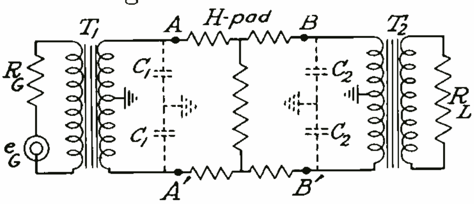


Fig. 12.--Example of balanced-to-ground circuit.

generator and load R_L are shown separated conductively from the balanced portion of the circuit, but this is not necessary if the impedance from either end of each to ground is equal to the impedance from the other end to ground. Note that the center tap of the secondary of T_1 and the primary of T_2 are grounded. Usually only one ground is necessary; it establishes the position of ground for the entire loop.

Consider corresponding points A and A'. The two connecting wires are generally equally spaced from ground (normally the shield surrounding them), so that the capacity to ground C_1 is the same for each. The value of R_G as reflected by T_1 to its secondary has in itself no particular reference to ground, hence the center tap definitely makes the impedance to ground from either end, half of the reflected value of R_G.

The use of an H-pad, which inserts equal resistances in the two sides of the circuit, as well as the fact that the two capacities of wires B and B' to ground are equal, and the primary of T_2 is center-tap grounded, all ensure that the impedance to ground from A and from A', or from B and from B', or from any other two corresponding points of the circuit, are equal. It is also clear from the figure that the voltage from A to ground equals that from A' to ground, and since the drop across AB is equal to that across A'B', the voltages from B to ground and from B' to ground are also equal.

An unbalanced-to-ground circuit is one that does not have the above balance of impedances and voltages to ground. There can be various degrees of unbalance, but normally an unbalanced-to-ground circuit means one that has one side grounded, as shown in Fig. 13. Note that the capacities of the

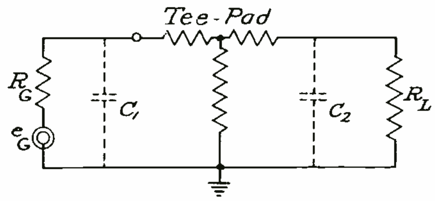


Fig. 13.--Example of unbalanced to ground circuit.

top wires to ground, C_1 and C_2, are operative and that the capacity of the bottom wire to ground is shorted out by the ground connection. The capacity loading is therefore greater because there is only one capacity across any part of the circuit instead of two in series, as in Fig. 12.

It will also be observed that in the unbalanced circuit a tee pad rather than an H-pad is employed, and that the series arms are *in the ungrounded or "high side" of the circuit.* This is an important point as regards cross talk.

Balanced circuits are in general preferable to unbalanced circuits because they are much freer from coupling to other circuits. In Fig. 14 is shown how a power line couples through stray capacity C to an unbalanced audio line, causing a power or noise current i_n to flow partly through

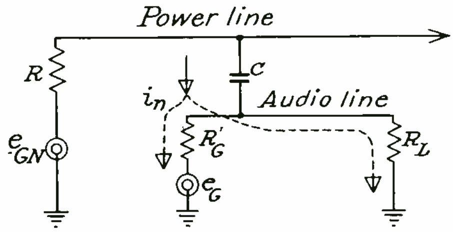


Fig. 14.—Noise pickup by unbalanced circuit.

R_G and partly through R_L. The resulting IR drop produces a noise voltage that passes on through the audio system. In Fig. 15 is shown the similar situation with respect to a balanced audio line. Here the power line couples through stray capacities C_1 and C_2 to both sides of the balanced line. The resulting noise currents i_1 and i_2 are nearly equal if C_1 and C_2 are nearly equal. The currents flow in opposite directions through the two halves of the primary of the output transformer, so that their magnetic effects cancel and no noise voltage appears across R_L, whereas the signal current flows in the same direction through the two halves of the primary and produces an additive effect.

The noise pickup by a balanced circuit is so much lower than that of an unbalanced circuit that practically all telephone lines are of the former type. Complete cancellation can occur only if $C_1 = C_2$ in Fig. 15. This is approximated by twisting the audio pair so that first one side of the circuit is closer to the power line and then the other, thereby averaging out the values of C_1 and C_2. This corresponds to the more careful method of systematic interchange known as transposition employed in telephone practice. Shielding surrounding the wires helps too, but in high gain circuits the appreciable resistance of a shield permits sufficient pickup of noise voltage to cause trouble in an unbalanced line.

In mixer circuits of the highest grade, particularly as regards freedom from noise pickup, balanced-to-ground inputs and outputs are preferred. This is particularly the case where low-level mixing is employed. Therefore the parallel

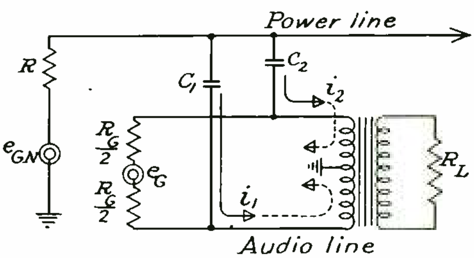


Fig. 15.—Noise pickup on balanced circuit.

type mixer is often used, because it lends itself particularly well to balanced operation.

Cross Talk Pickup.—Instead of a power line, the interfering circuit may be another audio circuit. If the signal level is high, then appreciable amounts of pickup will occur in the other circuit, and will be relatively high if the signal level of the other circuit is low. For that reason high-level and low-level circuits, such as microphone and Program Amplifier output circuits, are kept as far apart as possible in broadcast studios. Twisted pairs, well shielded, are almost invariably used, particularly for long runs in the building.

Another point to note is that if both circuits are balanced, then the pickup owing to stray capacity coupling will tend to cancel more completely, and if the circuits are of low impedance, capacity coupling will be less, although inductive coupling may be somewhat increased thereby.

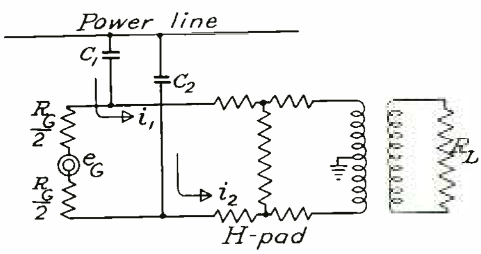


Fig. 16.--Use of H-pad in balanced circuit.

If a volume control is to be employed in a *balanced line*, then it should be of the *balanced type or H-pad*. Fig. 16 shows why this is so. The noise currents i_1 and i_2 will cancel if the impedance in each side of the circuit to ground is equal. But if a tee pad is employed such that the series resistors are in the top part of the circuit only, as shown in Fig. 17, then cancellation will no longer occur. Remember that i_1 and i_2 are determined mainly by the reactance of stray capacities C_1 and C_2 (Fig. 16). Now consider i_1 in Fig. 17. At point A it has a choice of two paths to ground: through $(R_2 + R_L/4)$, and through $(R_3 + R_L/4)$, and will divide between these two paths inversely as the ratio of the impedances of the two paths. It is clear that the division will in general be unequal through the two halves of the transformer primary.

Next consider i_2. At point B it has a choice of three paths to ground: $(R_3 + R_2 + R_L/4)$, $(R_G + R_1 + R_2 + R_L/4)$, and through $R_L/4$. It is even more evident that i_2 will not divide equally between the two halves of the transformer primary. It may be that for some particular pad setting

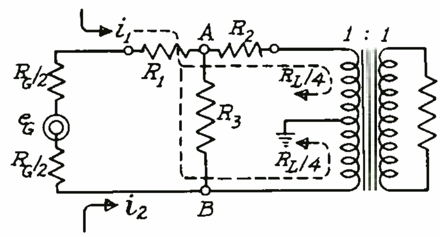


Fig. 17.--Use of tee-pad balanced circuit.

the division of i_1 and that of i_2 will just balance in the primary of the transformer, but in general such balance will not occur, so that noise pickup and cross talk will take place.

Another factor is that the tee pad will not provide the same attenuation at all frequencies. This is shown in Fig. 18 (A), (B), and (C). The method given here should prove useful to the engineer in the analysis of many otherwise inexplicable causes of hum, cross talk, and even feed back in an audio system or any other a.c. circuit. Refer to Fig. 18 (A). A current i_t flows from the generator, which may be the output of a pre-amplifier. Capacitors C_1 and

C_2 represent the capacities of the lines to ground on either side of the tee pad. At A the current encounters the series impedance of the pad. Some of it flows through as i_1, the rest is diverted as i_4 through the top capacitor C_1.

of i_3 flows through the transformer primary and induces a voltage across R_L. (It is understood that R_L could be a center-tapped resistance placed directly in circuit with the pad in place of the transformer, and that furthermore the transformer can have

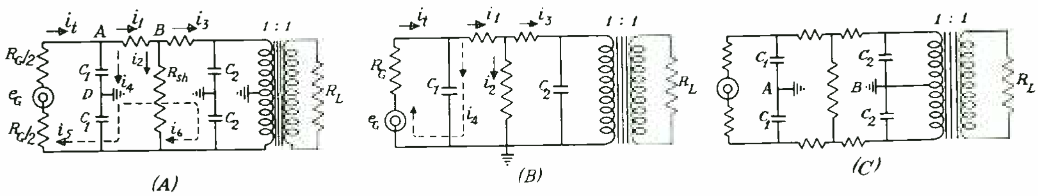


Fig. 18.—Effect of improper and proper connection of pad on frequency response.

At B some of i_1 flows down through the shunt resistor R_{SH} of the pad as i_2, the rest flows through the other series arm as i_3. If the pad is set for high attenuation, then R_{SH} will be very small, so that i_2 will be the major portion of i_1, and i_3 will be only a small fraction of i_1 even though the resistance it flows through is not very large. While i_4 through C_1 is normally small compared to i_1, it can nevertheless be comparable to i_3 in value if the attenuation of the pad is high, especially at the higher frequencies, where the reactance of C_1 may be relatively low. Note that i_3 is the normal current reaching the load through the pad, while i_4 is a stray current flowing through the stray capacity of the line.

Current i_3 has a choice of C_2 or the reflected value of R_L (primary of the transformer). The latter impedance is much lower than the reactance of C_2, so that most

a turns ratio other than unity as shown.)

From the above considerations it is clear that line capacity C_2 *after the pad* and shunting the load has very little effect, whereas line capacity C_1 *preceding the pad* may have an appreciable effect, as will be shown. Thus consider i_4. At point D it has a choice of two paths: through the high reactance of the lower C_1 back to the generator as i_5, or via the dotted line path through ground and the lower half of the transformer primary as i_6. Since the latter is the lower impedance path, i_6 is nearly equal to i_4, and is therefore comparable to i_3, and hence will contribute an appreciable increase to the voltage induced in the secondary across R_L. Since this effect is more marked at the higher frequencies, it is apparent that a frequency response test will show a rising characteristic. However, this effect is appreciable only at high attenuation settings of the pad, and this is not the case in normal operation. But what is important is that as the pad is set to infinite attenuation, appreciable amounts of high-frequency signal will by-pass the pad and get to the load; i.e., the pad will not have infinite attenuation at the higher frequencies. In particular, an especially loud sound impinging on the microphone may produce considerable output into a mixer from a microphone considered dead; i.e., cross talk will be experienced.

In Fig. 18(B) is shown the correct use of a tee pad, namely, in an unbalanced circuit. Note that the component i_4 of i_t has only one path, through C_1 and back to the generator. It cannot get into the output load, so that only i_3 flows through the latter. Since the magnitude of i_3 is determined by the attenuation setting of the pad and is moreover independent of frequency (being determined simply by the ratio of resistances in the pad), a flat frequency response at all pad settings can be expected, and what is possibly even more important, no cross talk need be experienced.

In Fig. 18(C) is shown the correct use of an H-pad, namely, in a balanced circuit. From the balanced nature of the circuit, ground points A and B are at the same potential, so that there is no tendency for current through C_1 to get past the H-pad and into the load. Thus a flat frequency response at all pad settings and freedom from cross talk can be expected.

The above considerations are not only important in the case of a mixer, but in setting up test equipment for making a frequency run on an amplifier, etc. Many peculiar results have been obtained because care was not exercised in choosing and arranging the equipment.

Cross Talk in a Mixer.—From the above it is clear that if the pre-amplifier outputs are balanced to ground, then the fader pads in an associated parallel-type of mixer should be of the H-type, and if they are unbalanced to ground, then the fader pads should be of the tee type. All must be of the same type, H or tee, since all are connected essentially in parallel. A further refinement in the case of the balanced parallel mixer is that the compensating resistances R_p should be divided in half and placed in each side of the circuit. This is shown in Fig. 19.

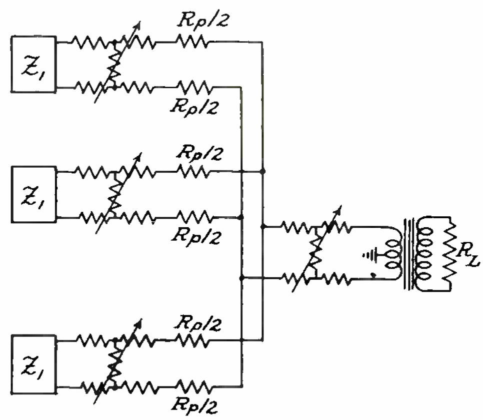


Fig. 19.—**Symmetrical Arrangement of components in a balanced parallel mixer.**

As an example of cross talk in a mixer, consider the five-position series type shown in Fig. 20.

For simplicity, the pre-amplifiers are shown merely as generators having an internal impedance Z_1, and the top fader tee pad is shown set for infinite attenuation, in which case its shunt arm is zero, and it reduces to the form shown. The capacities of the leads to ground are indicated by the condensers C. The center position #3 is in a symmetrical position with regard to the rest of the circuit (similar to the output impedance Z_2) and hence is shown as having an H rather than a tee fader control. No master Gain Control is shown as this does not affect the impedance looking towards the output nor the occurence of cross talk.

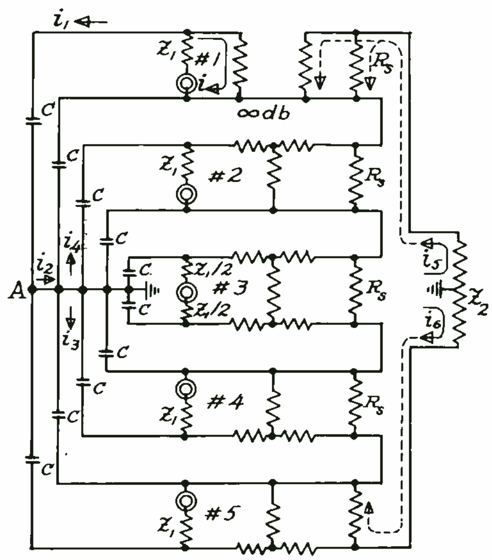


Fig. 20.—Cross talk in a five position series mixer.

Cross talk is most pronounced from the end positions #1 and #5. Consider Position or Station #1. Current i is the normal current that flows through the pad. For the infinite attenuation setting, none of this current gets to the load Z_2.

Current i_1 is a capacity current that produces cross talk. It flows through C to point A. Here it has a choice, either through the various line capacities C as i_3 and i_4, or through ground to Z_2 as i_2. Since the latter path is of much lower impedance, i_3 and i_4 will be a negligible fraction of i_1, and i_2 will nearly be equal to i_1.

Now consider the path of i_2. At Z_2 it can flow upward as i_5 to get back to the lower side of generator #1, or downward and around through the other fader positions as i_6. The latter path has clearly a higher impedance than the former, so that i_6 will be less than i_5. The difference becomes greater as the number of positions is increased; on the other hand it is zero if only two positions are involved. But if i_5 exceeds i_6, there will be a net unbalance in the voltage drops across the halves of Z_2, and hence a net voltage developed across the ends of Z_2. This net voltage is the cross talk voltage that appears in the output Z_2 from generator #1 even when the latter's volume control is set for infinite attenuation.

From what has just been said, it is clear that a two position series mixer will have no cross talk, but that a series mixer having a greater number of stations will have cross talk, particularly from the stations off center and particularly at the higher frequencies. For this reason a series mixer of more than two stations is not recommended, while the series-parallel mixer in which only two stations are connected in series in each group will be satisfactory.

Parallel type mixers are free from cross talk for any number of positions, provided they are designed as indicated previously. The same is true of the bridge type mixer, and of the modified series-parallel mixer.

Summary of Mixer Design.-- From the foregoing it is possible to prepare a table or summary of recommended mixer design. It is based on the assumption that minimum insertion loss is desired as well as freedom from cross talk.

TABLE OF MIXER DESIGN

INPUTS	NO. OF STATIONS	TYPE	OUTPUTS	REMARKS
Balanced	Any Number	Parallel	Balanced	Least noise pickup.
Unbalanced	Odd Number	Parallel	Unbalanced	Loss equal to Mod. S-P having one more station.
Unbalanced	Even Number	Modified Series-Parallel	Balanced	Less loss than parallel type.
Unbalanced	Two	Parallel	Unbalanced	
Unbalanced	Two	Series	Balanced	
Unbalanced	Four	Bridge	Balanced	Min. Loss of 6 d.b. All impedances equal.
Unbalanced	Four	Series-Parallel or Parallel Series.		Used, but has greater loss than Bridge type.

The Inputs and Outputs are described as they must inherently be. For example, if one wishes to use a five or six position mixer, (odd or even value of N), and the inputs are inherently balanced to ground, then the table, Row 1, shows that a parallel mixer is required, and that the output will be inherently balanced to ground.

Of course, isolation transformers can be employed to convert the balanced inputs to unbalanced inputs, and the same is true for the output. The transformers can also transform the required, possibly non-standard value of impedance into a standard value. The above table refers to a design in which such transformers are not employed.

Now suppose that a mixer of five positions, unbalanced, is desired. Since five is an odd number, Row 2 shows that a parallel mixer is still required, but that now the output will also be inherently unbalanced. On the other hand, if a mixer of six positions, unbalanced, is desired, then Row 3 shows that a modified series-parallel mixer is specified, rather than a parallel type. This, however, is simply on the basis of reduced insertion loss. As far as cross talk is concerned, a parallel mixer would be just as satisfactory. Further practice in using this table will be obtained from the examination problems.

BOOSTER AMPLIFIER.—One point should be noted in connection with the insertion loss. The values given here are the ***minimum***; i.e., with the volume controls set at zero attenuation. In some types of controls, such as those employing a ladder network, there is a minimum loss of about 6 db even when the pad is "wide open". Furthermore, in normal practice the controls are never turned to minimum attenuation, but operate at about three-quarters of maximum volume so as to afford latitude in the monitoring of the program. For this reason the actual or normal insertion loss may exceed the above minimum values by as much as 10 db or more. In some of the older types of mixer that were not designed properly, much higher minimum insertion losses were encountered.

In all cases, the insertion loss increases as the number of stations is increased. Hence it may be found that the output has dropped to such a low level that the signal-to-noise ratio is too low. In such a case it is found desirable to insert an amplifier, known as a Booster Amplifier, between the outputs of the faders and the Master Gain Control, in order to obtain a favorable signal-to-noise ratio even after passing through the latter.

ATTENUATION PADS.—The noise referred to is not only that of

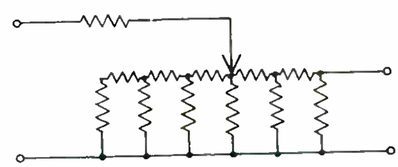


Fig. 21.—Ladder pad, unbalanced type.

thermal agitation, but noise developed in operating the volume controls. A tee pad consists of three separate rheostats mechanically connected, so that three arms and points of contact are involved, whereas an H-pad has six arms. The latter therefore tends to be noisier in operation, and is a more complicated structure. The ladder type of network, illustrated in Fig. 21, is equivalent to a tee pad, but has only one con-

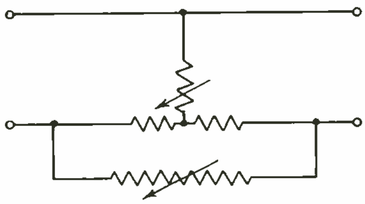
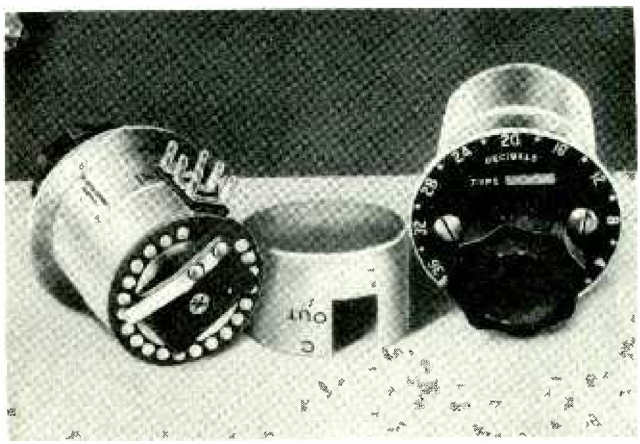


Fig. 22.--Bridged tee pad.

tact arm. It has, however, a minimum insertion loss of about 6 db, and moreover does not afford quite as perfect an impedance match in both directions as does the tee pad, particularly near the extreme or end positions.

In the form equivalent to the H-pad it has two contact arms, but even this is less than the number required for the tee pad, and one-third the number required for an H-pad. Another type of pad that has all the desirable qualities of the tee pad but requires only two arms, or four arms in its balanced form, is the bridged-tee pad. This is shown in Fig. 22. The shunt and bridging arms are the variable resistances. One possible disadvantage as compared to the ordinary tee pad is that in the latter, the three rheostat arms can be electrically connected together, which possibly makes for a stronger mechanical assembly, whereas in the case of the bridged-tee pad, the two arms must be insulated from one another, as is evident from Fig. 22. As an example of a pad, Fig. 23 shows a photograph of a



(Courtesy Daven Co)

Fig. 23.--Ladder network.

ladder pad, and Fig. 24 shows a tee pad, while Fig. 25 shows a mixer console.

Courtesy Daven Co.

Fig. 24.—Tee pad.

The noise generated in a well-designed pad may be -120 db to -100 db (0 db = 6 mw.), depending upon the number of contacts. The noise is owing to contact potentials developed between the arm and the studs it passes over. Such contact potentials can be made very small by using the same metal for the arm and for the stud, but unfortunately similar metals do not wear well when rubbing against one another except in special cases. It has been found that if a very thin film of beryllium is deposited on the surfaces, the wear is greatly reduced. Also a special oil sold under the name of Davenoil has been found to be effective in reducing the wear.

Fader pads are often designed to increase the attenuation uniformly with rotation of the dial, say in 1.5 or 2 db steps, until the last contact is reached. Then an amount of attenuation is inserted about equal to all that previously inserted. Thus the music dies out gradually, and then more abruptly, but by that time the level is so low that the abrupt additional change is not particularly notice-

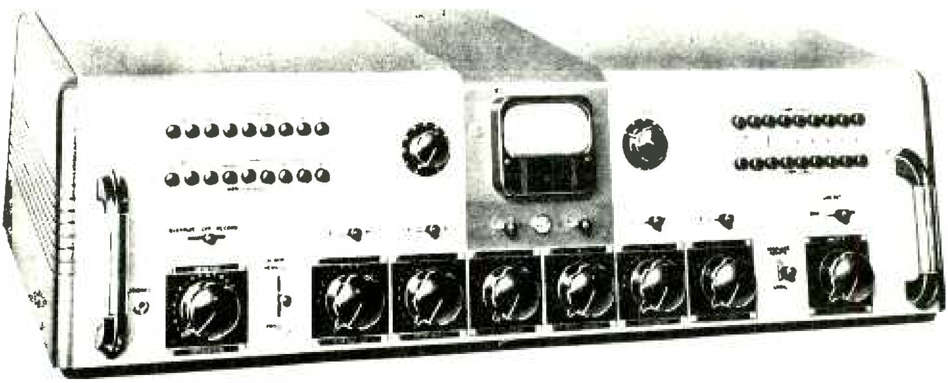


Fig. 25.--Representative example of a mixer console.

able.

Detents can in general be added to the pad so that the contacts can be told by a click, but where rapid variations in either direction are desired—as is usually the case in monitoring,—a smooth control may be preferred. Finally, the casing around the pad not only protects the contacts from dust and dirt, but acts as an electrical shield as well.

ELECTRONIC MIXING.—It is also possible to mix two signals by means of electronic methods. Specifically, in a two-position mixer of this type, two tubes are employed. Each signal comes in on its respective grid, and the plates of the two tubes are connected together in parallel to a common load resistor.

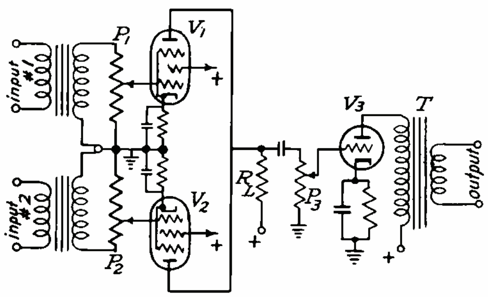


Fig. 26.—Two-position electronic mixer.

The circuit is shown in Fig. 26. Tubes VI and V2 have a common load resistance R_L, hence their two amplified signals are superimposed or mixed in R_L. The signal then is further amplified by V3, and the output is obtained via the tube-to-line transformer T.

Note that the two fader positions are ordinary potentiometers P1 and P2, and that the Master Gain Control is potentiometer P3. Thus single-arm potentiometers may be used instead of the more complicated types of pads required for the resistance network type of mixer.

Alternatively the gain may be controlled in any of the tubes by using supercontrol tubes, in which the transconductance, and hence the

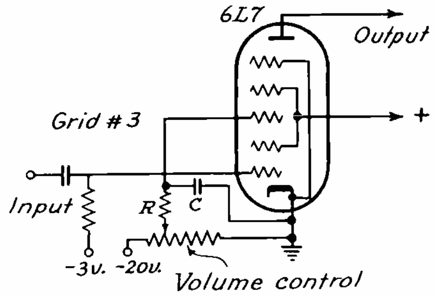


Fig. 27.—Use of bias control to vary gain of supercontrol tube.

gain, is varied by varying the bias on the grid of the tube. The grid may be the control or some other grid in the tube. In the case of the 6L7 tube, for example, the signal may be impressed on grid #1, the control grid, and the adjustable bias on grid #3. As indicated in Fig. 27, the bias on the control grid is maintained constant, but the bias on grid #3 can be varied over a wide range. The transconductance can accordingly be varied from about 1,100 μmhos down to 5 μmhos. It is also possible to apply the signal to grid #3, and the bias to grid #1.

Since control is of a d.c. (bias) circuit, the leads can be as long as desired between the potentiometer volume control and the supercontrol tube. This means that the tube can be located in the rack, and the volume control on the console. There are, however, certain disadvantages connected with the electronic mixer as well as the advantages of simpler volume controls, that of amplification rather than attenuation occurring in this type of mixer, and the like. In the first place, a vacuum tube is not as reliable as an attenuator pad, and hence more trouble-free service can be expected from the ordinary resistance-type mixer than from the electronic type.

A second disadvantage is that of somewhat greater distortion owing to the additional vacuum tubes introduced into the system. This may be even more marked if the type of volume control shown in Fig. 27 is employed. Suppose a strong signal is impressed upon the tube. To keep this from overloading the rest of the system, the bias on the tube is increased to reduce its gain. But this brings the path of operation onto the more curved part of the tube characteristic, where distortion is more pronounced. It is therefore important to insure that the signal level is sufficiently low at this point so that undue distortion does not occur.

As indicated in Fig. 26, pentode tubes are preferable for electronic mixing, because the load impedance to either tube is not merely R_L, but R_L shunted by the r_p of every other tube connected to R_L. For a multi-position mixer, this shunting effect may be considerable unless r_p for each tube is high, as in the case of the pentode. However, also note that the plate currents of both V1 and V2 (Fig. 26) flow through R_L. If many tubes are involved, the drop in R_L may be excessive and insufficient plate voltage will be applied to each tube unless the supply voltage is made unduly high, or R_L is reduced. Reduction in R_L, however, results in reduced gain.

Fig. 28 shows a circuit variation that attempts to avoid this difficulty. Tubes V1 and V2 have separate plate load resistors R_{L1}

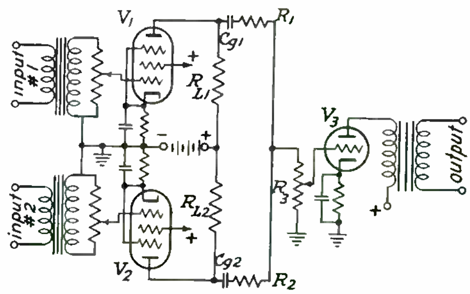
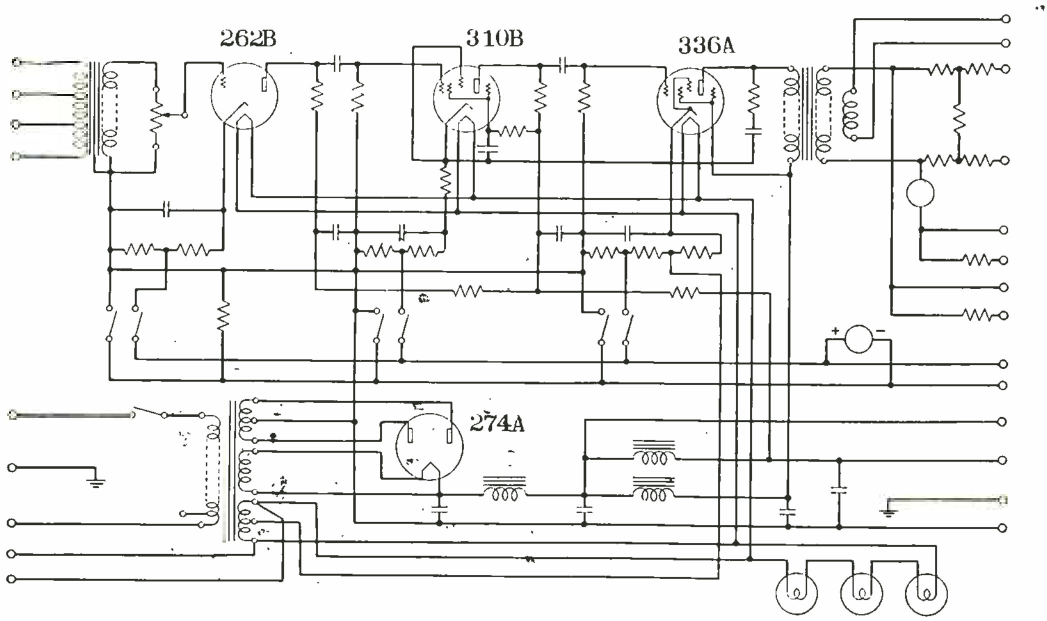


Fig. 28.--Another electronic mixer design.

and R_{L2}, so that each can be relatively high and yet not produce excessive drop in the plate voltage. Coupling is now in the grid circuit via C_{g1} for V1 or C_{g2} for V2. For either tube, following its coupling condenser, there would be the grid resistor R_3, paralleled by the other coupling condenser in series with the other plate load resistance. The latter shunt circuit would provide a variable reactive load with frequency and spoil the frequency response. To prevent this, compensating resistors R_1 and R_2 are inserted between C_{g1} and C_{g2} and R_3. Thus, if R_3 is fed from V1, the impedance paralleling R_3 is not C_{g2} and R_{L2}, where R_{L2} and C_{g2} are both low, but C_{g2} and $(R_2 + R_{L2})$, where the latter resistance is so high (owing to R_2) as to mask the variable series reactance of C_{g2}. Thus the impedance shunting R_3 is essentially $R_2 + R_{L2}$ when V1 is feeding signal, and $R_1 + R_{L1}$ when V2 is feeding signal; i.e., a pure resistance in either case of practically constant value with frequency. However, it is necessary that R_1 and R_2 be about twice as large as the reactance of C_{g2} or C_{g1} at the lowest frequency it is desired to amplify, and hence that these be comparable to R_3 in value. This results in an appreciable voltage divider action between R_1 and R_3 or R_2 and R_3, and the gain is correspondingly reduced. The greater the number of stations to be mixed, the higher must the compensating resistors be, and hence the lower will be the gain. Both types of circuits are successfully employed, but generally for no more than four inputs. It is to be noted in passing that the circuit of Fig. 26 is somewhat more free from phase shift, and hence is preferred to that of Fig. 28 in video (television) amplifiers for mixing purposes.

With reference to the use of a supercontrol tube, one difficulty arises in that a change in bias not only changes the transconductance and hence the amplitude of the signal (a.c.) component of the plate current, but also the d.c. component. A quick change in the latter, as when the bias rheostat is suddenly changed, produces an amplified effect through the system and results in a "thump" in the loudspeaker. To minimize this effect, a time delay is inserted in the grid circuit. This is shown in Fig. 27 as R and C; these delay the change in bias with regard to the grid of the tube and produce a more gradual change. Too high a time constant, however, will make the volume control sluggish and unsatisfactory for monitoring purposes. A more satisfactory solution is to use a pair of tubes in push-pull for each position, with their supercontrol grids, however, connected in parallel to the bias potentiometer. In this case the change in bias appears *in the same polarity* to the two tubes, while the signal voltages to the control grids are *180 degrees out of phase* (normal push-pull connection). Owing to the push-pull connection of the plates, the signal currents of the tubes produce *additive* effects in the output circuit, whereas the "thump" currents or changes in d.c. component are in phase and cancel in the output circuit. The success of this method depends upon the degree of balance between the two tubes, and the disadvantage in any case is the fact that two tubes are required for each position instead of one.

Electronic mixers are in general used in small station equipment where simplicity and compactness and economy are of paramount importance, and where possibly two fader positions are all that are required to be mixed simultaneously. In such equipment the faders are often mounted on the amplifier rack instead of a separate console, and monitoring accomplished at the rack itself. Electronic mixing is also often employed in public address systems, because the mixing operations are of a more rudimentary nature and monitoring is seldom



(Courtesy Western Electric Company)

Fig. 29.--Western Electric 105A Program Amplifier.

required here. In large broadcast and high-quality recording systems, however, the resistance type of mixer is still preferred because while it is more expensive and bulky, it nevertheless is more flexible in meeting the requirements as to number of positions, is probably more reliable, and is somewhat more free from distortion.

PROGRAM AMPLIFIER.—In Fig. 29 is given the schematic wiring diagram of a Program Amplifier. It will be observed to be a self-contained completely a.c.-operated unit, and consists of three stages, the first two of which are resistance-coupled, and the last stage of which has a tube-to-line transformer to feed a 600-ohm (or 500-ohm) line. A 6 db line isolation pad is incorporated in the output. In addition, a second output winding of greater step-down (to about 40 ohms) is provided to deliver a voltage approximately 20 db below the main output. This is intended to be connected to a Western Electric 94 Type (or equivalent) amplifier for monitoring purposes.

It will be observed that the first stage employs a triode tube (WE 262B), and that the last stage employs a pentode tube (WE336A). The feedback from the plate of this tube to the cathode of the preceding stage not only reduces the distortion, but also the internal impedance of the output tube, thus enabling it to match the output transformer. The second stage has a WE 310B pentode tube, which affords high gain in a resistance-coupled stage. Also note that the input transformer is designed to operate from a 600-ohm line (also 500-ohm line), but that it is tapped so that it may be operated from a 30-ohm line if desired.

The amplifier gain is approximately 70 db (including the 6 db output pad) and the gain control across the secondary of the input transformer has nineteen 2 db steps of attenuation plus an "OFF" position. It comes regularly equipped with a volume indicator (bridged across the input of the 6 db isolation pad) and plate current meter which may, however, be omitted where desired.

In some designs of broadcast amplifiers the volume indicator is also used as a plate current meter by having a switch connect it across the cathode bias resistor of each tube. This indicates not only whether bias is being provided for the tube, but also, from the voltage drop across the cathode resistor, whether the normal plate and screen currents are flowing. It affords a quick check as to whether or not the tube is in good condition.

LINE AMPLIFIER.—The Program Amplifier shown above was designed to feed the outgoing telephone line directly. While this is quite normal practice, the alternative method mentioned previously whereby the Program Amplifier feeds an Audio Bus, and various subsequent amplifiers bridge off this bus, is also used extensively.

In such a case, a Line Amplifier is employed to feed each outgoing telephone line. An example of such an amplifier is shown in Fig. 30. As will be noted from the figure, the unit is completely a.c. operated and self-contained. There are two stages: both employ pentode tubes that are triode-connected so as to enable them to be transformer-coupled to one another.

The input transformer is tapped for 500- or 20,000-ohm input. It can therefore function as a regular input or as a bridging transformer, as desired. This can be seen from the following considerations: Potentiometer R-1 across the secondary reflects to the primary terminals a resistance that depends upon the turns ratio. The number of turns between terminals 1 and 6 is clearly greater than that between 2 and 5, hence the step down from the secondary to 1-6 is less than that to 2-5.

Hence the same resistance R-1 appears as high as 20,000 ohms when viewed from terminals 1-6, and as 500 ohms when viewed from terminals 2-5; it is stepped down to a lower value in the latter case. If terminals 2-5 are connected to a 500-ohm line, then they furnish the necessary 500-ohm termination, and a second amplifier whose input impedance is 500 ohms could not be connected because the joint impedance would then be 250 ohms, which would be a mismatch to the line.

On the other hand, if terminals 1-6 are connected to a 500-ohm line, the high impedance of 20,000 ohms is connected thereby across it, and one could connect $20{,}000 \div 500 = 40$ such amplifiers across the line before the joint impedance would be reduced to 500 ohms. Or, as is actually done in practice, a 500-ohm resistor can be connected across the line (called an Audio Bus) and then as many as five 20,000-ohm input amplifiers can be connected across the bus without materially reducing the initial 500-ohm impedance.

These amplifiers are known as bridging amplifiers, and the input transformer, when terminals 1-6 are employed, is known as a bridging transformer. From the foregoing discussion several things are apparent:

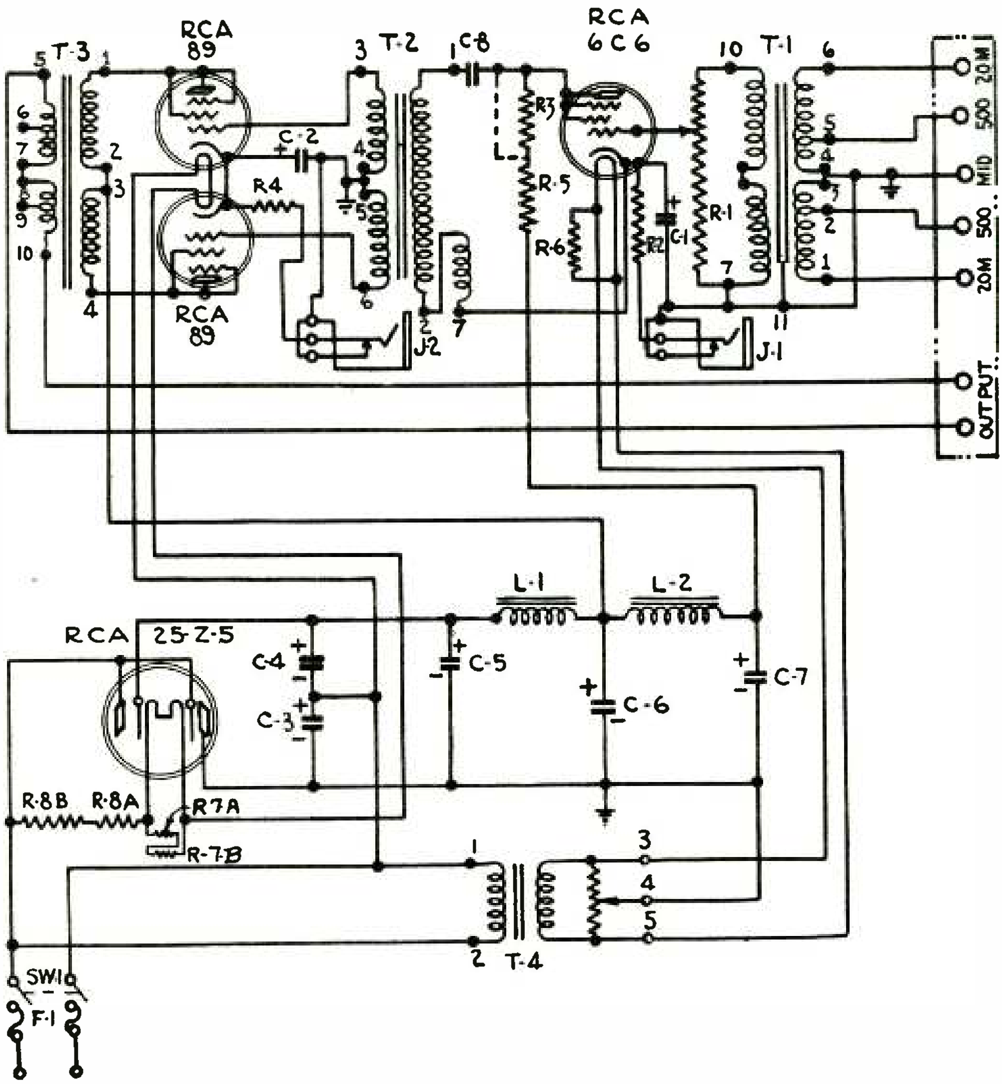
1. A bridging amplifier differs from an ordinary amplifier only in that a bridging input transformer is employed, and the latter is merely a transformer that has an input impedance very much higher than the source impedance feeding it. For example, an ordinary 500-ohm input transformer will be a bridging transformer when fed from a 15-ohm line.

2. The same amplifier and input transformer can be of the normal or of the bridging type, depending upon which taps on the primary of the input transformer are employed. The above 55-A Line Amplifier illustrates this fact very clearly.

3. The step up of the input transformer when the high-impedance or bridging terminals are employed is much less than when the lower normal impedance terminals are employed. This in turn means that the gain of the amplifier when used as a bridging type is lower than when used in the normal fashion, since for a given voltage generated by the source, there is less voltage at the grid of the first stage.

For this reason the normal impedance connection is preferred for an amplifier *if it is to be the only load connected to the line.* But where several independent outputs are desired, the lower-gain bridging connection is clearly preferable. In any amplifying system, it is desirable to use bridging amplifiers near the high-level end of the system, since then much less additional gain will be required in *each* of the bridging amplifiers, and the preceding amplifiers function essentially as a common *pre-amplifier* for *all* the bridging amplifiers.

In view of the above, the Pro-



(Courtesy RCA)

Fig. 30.—RCA Type 55-A Line Amplifier.

gram Amplifier is normally of the matched input impedance type having high gain, and bridging is employed at its output, as in the case of the above Line Amplifier. Such a system is more flexible and economical of equipment, and permits bridging amplifiers to be connected or disconnected from the audio bus without fear of impedance mismatch.

The output impedance is the normal line value of 500 ohms, or 250 ohms if terminals 6 and 9 are employed instead of 5 and 10, and the maximum gain is 46 db (for 500-ohm input). A simple calculation enables the gain as a bridging amplifier to be calculated. The increase in gain for the matched connection as compared to the bridging connection is

$$20 \log \sqrt{Z_b/Z_m}$$

where Z_b is the bridging input impedance and Z_m is the lower matched impedance. For the Type 55-A amplifier this is $20 \log \sqrt{20000/500} = 20(.8011) = 16$ db. Hence the gain of the Type 55-A when used as a bridging amplifier is 46 - 16 = 30 db which is ample to feed the line because the output of the Program Amplifier is at a relatively high level.

The volume control in the secondary of the input transformer provides a gain reduction of 38 db in 2 db steps. The maximum gain can be reduced by 10 db, if desired, by connecting coupling condenser C-8 to the terminal between R-3 and R-5 as per the dotted line shown.

One interesting feature of this unit is the method of obtaining plate voltage. Instead of using a step-up power transformer a voltage-doubling circuit consisting of condensers C-4 and C-3 and the 25-Z-5 rectifier tube is employed directly off the 120-volt line. The elimination of the power transformer minimizes hum pickup by the input transformer.

Further reduction in hum output is obtained by means of the potentiometer connected across terminals 3-5 of the filament transformer T-4. To adjust, short the 20,000-ohm input terminals to the MID (ground) terminal, connect a headphone to the output terminals, and adjust the potentiometer until minimum hum is heard. Ordinarily this will occur over an appreciable range of adjustment. The potentiometer arm is then set to the center of this range. In addition, the filament transformer should be placed as far from the input leads as possible. Note that this transformer is required for the heater of the 6C6 tube; the heaters of the 2-89 tubes are in series with the filament of the 25-Z-5 and R-8A and R-8B resistors across the 120-volt line.

MONITORING AMPLIFIER.--The monitoring amplifier is also of the bridging type, but differs from the Line Amplifier in that it has inherently higher gain, since its output level must be great enough to actuate a loudspeaker instead of merely feeding a telephone line. A power output stage is therefore indicated.

In Fig. 31 is shown a monitoring amplifier designed either to bridge across a line or to match a 500-ohm impedance. By using a tapped primary input transformer, the same amplifier may be used not only for monitoring purposes, but also in an audition circuit where it may be fed directly from the

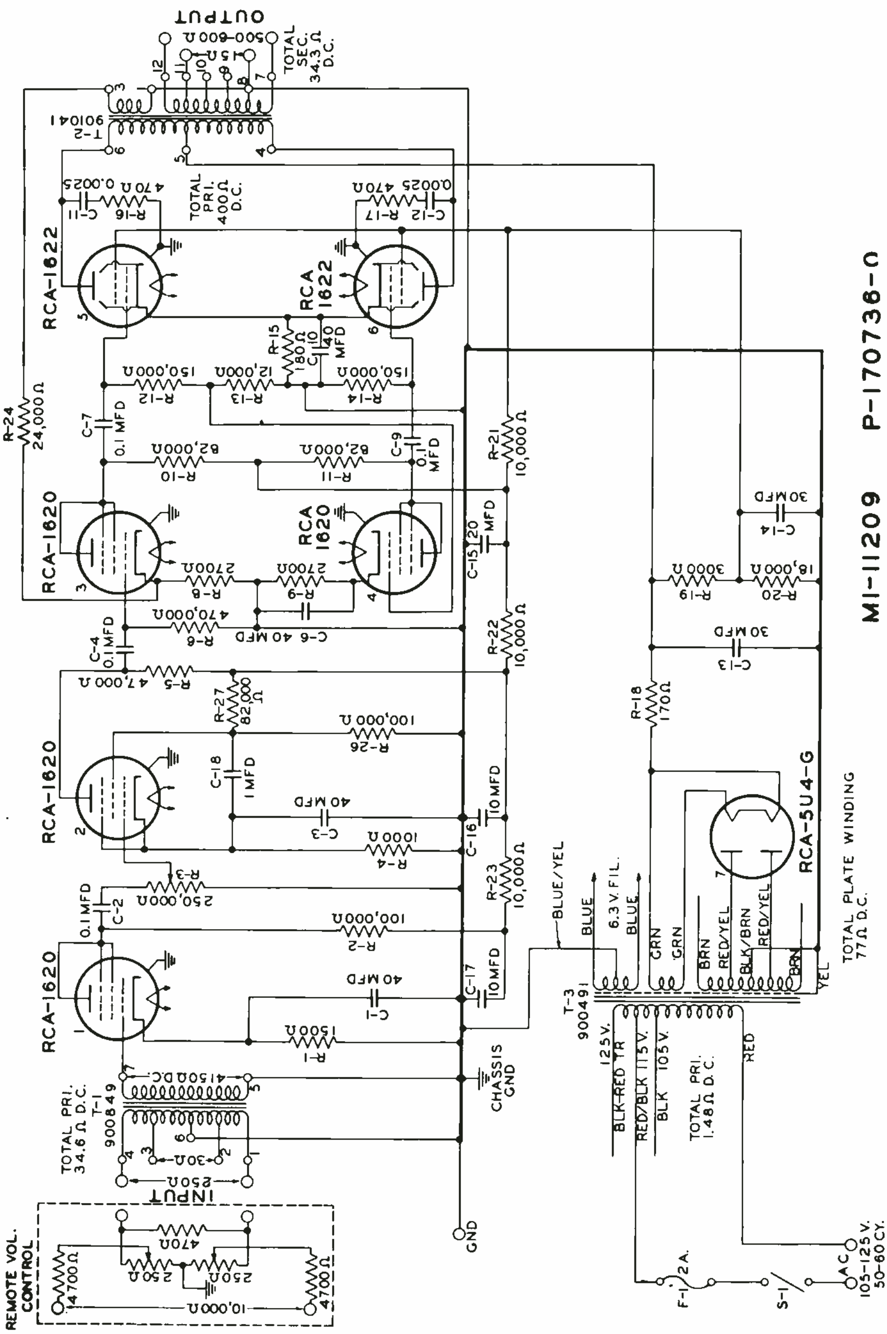


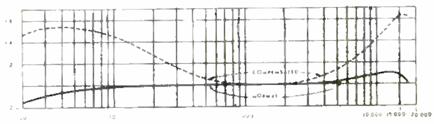
Fig. 31.—Schematic diagram-Type 82-C monitoring amplifier. (Courtesy RCA)

mixer output of a studio, or from a transcription turn-table, etc. This permits the manufacturer to employ the same stock items for broadcast, public address, recording, and other equipment.

As indicated in the figure, four stages are employed, of which the final or power output stage uses two Type 1622 beam power tubes (special 6L6 tubes) in push-pull. Observe the tertiary (third) winding on the output transformer. This is employed to provide inverse feedback to the cathode of the preceding stage. This not only helps to produce a flatter frequency response and reduce harmonic distortion, but provides a lower internal output or source impedance to the monitor loudspeaker, so that its transient response is improved owing to the greater damping provided.

The first two stages are resistance-coupled, and the next stage is a phase inverter using a Type 1620 tube to provide push-pull excitation for the final stage. The output transformer has secondary taps for 5, 7.5, 15, 250, and 600 ohms, so as to be able to feed one, two, or three 15-ohm loudspeakers or a 600-ohm line.

When used as a bridging amplifier a 10,000-ohm volume control is interposed between the line and the outer input terminals. This control can be located as far away as two or three hundred feet from the amplifier. A further feature is that by adding an R-C network in the plate circuit of the second stage, a 5 db boost at 60 c.p.s. is possible, and by adding an R-C network in the cathode of the third stage, a 6 db boost at 15,000 c.p.s. can be obtained. In this way compensation can be made for deficiency in output of the monitor loudspeaker at the low and high frequencies. Finally, it is to be noted that the unit is completely a.c. operated, but that hum pickup



(*Courtesy RCA*)

Fig. 32.--Frequency response of 82-C1 Monitor Amplifier.

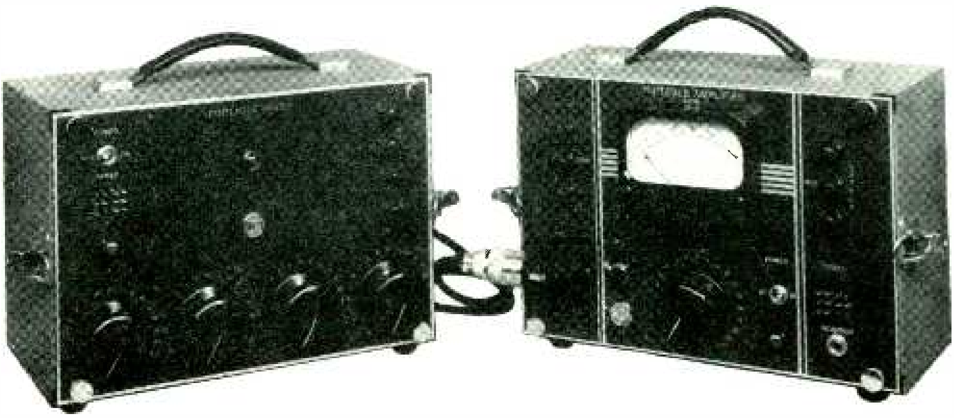
is kept to a minimum by a multiple-case shielded input transformer, and that plug-in electrolytic capacitors are employed to simplify servicing the equipment. The normal and compensated frequency response curves are shown in Fig. 32.

REMOTE PICKUP EQUIPMENT.--Many programs originate outside the studios. Examples of this are pickups at baseball and football games, meetings and banquets, night-club entertainment, particularly dance music, etc. These are called remote and sometimes "nemo" (pronounced neemoh) pickups.

It is clear that essentially an abridgement of the studio equipment is required at the remote point; one or more microphones, a mixer unit, combination program and line amplifier, and some monitoring means such as a pair of headphones. The mixer and amplifying equipment is generally built into one cabinet, and the two main requirements are lightness and portability.

In the past this was hardly the case, but considerable progress has been made and modern units fulfill the above two requirements very satisfactorily.

In Fig. 33 is shown a picture of such equipment, and in Figs. 34 and 35 are given schematic wiring diagrams of the two units employed. The OP-7 Mixer Pre-amplifier enables four microphones of either 250- or 30-ohms impedance to be connected, and each microphone signal is amplified by an RCA-1G20 triode-connected tube (see Fig. 34). The outputs of the four triodes can then be adjusted by the volume controls shown, and combined in parallel to be fed to controls are on the OP-7 panel, and the master gain on the OP-6 unit, as is evident from Fig. 33. The best signal-to-noise ratio is obtained by operating the active fader volume controls at or near maximum gain. The microphone cables plug into receptacles at the rear of the OP-7 unit, and the output of the OP-7 unit is normally connected to the OP-6 unit by means of a cable connecting plugs at the rear of the units. An additional OP-7 pre-amplifier unit may be connected in parallel with the first by means of the output



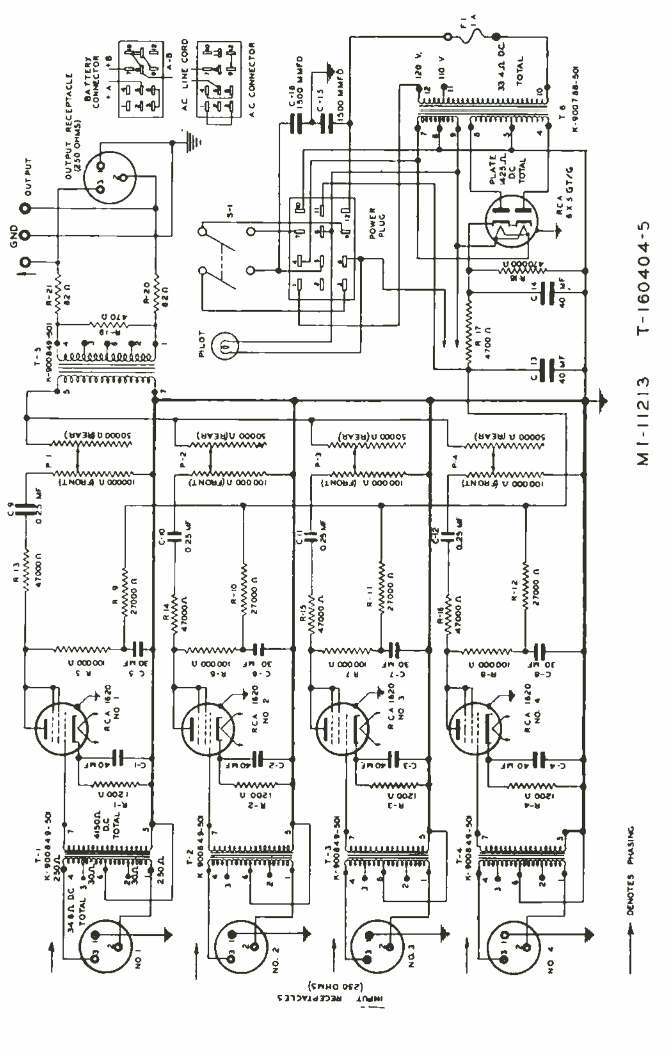
(Courtesy RCA)

Fig. 33.--Type OP-6/OP-7 remote pickup equipment.

the OP-6 Portable Remote Amplifier. Reference to Fig. 35 shows the latter to be a high-gain three-stage resistance coupled amplifier employing 3 RCA-1G20 pentode-connected tubes. A battery box (not shown) is also furnished, so that the equipment may be operated either on a.c. or d.c., as desired.

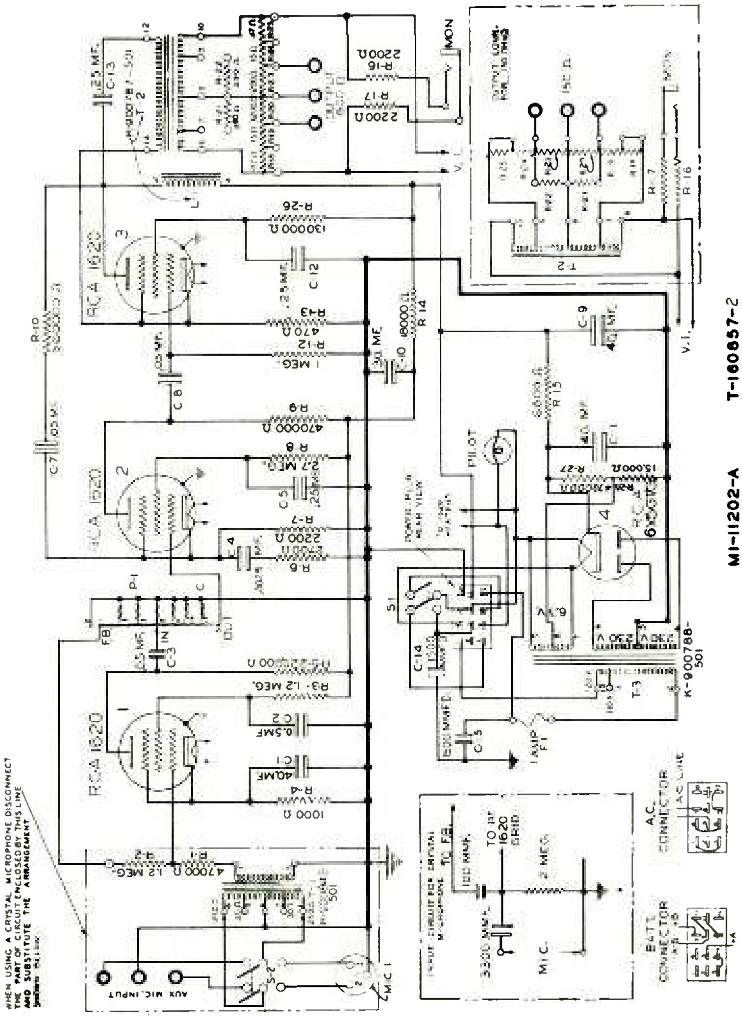
The mixer is divided between the two units: the fader volume screw terminals on the front panel of each.

Reference to Fig. 34 reveals some interesting circuit features of the OP-7 pre-amplifier. Note first that the secondaries of the input transformers are unloaded. As mentioned in the technical assignment on microphones, this gives a better signal-to-thermal-noise ratio that the matched (properly loaded)



(*Courtesy RCA*)

Fig. 34.--Schematic diagram of OP-7 portable mixer pre-amplifier.



(Courtesy RCA)

Fig. 35.—Schematic diagram of OP-6 portable remote amplifier.

condition will afford. In turn, this indicates the desirability of having the volume control in the plate side of the pre-amplifier tube; not only is input loading obviated, but any noise developed by the volume control during operation is relatively smaller compared to the amplified microphone signal on the plate side of the tube.

The volume controls are of a modified tee-pad construction. Such a pad can maintain a constant impedance *in one direction only when* varied. The four pads shown connected in parallel are arranged so that their output terminals (5,7) present a constant resistive impedance to the tube-to-line transformer T-5 regardless of their settings. This is because the frequency response of a transformer is sensitive to variations in source impedance.

The 1620 pre-amplifier tubes, on the other hand, each face a somewhat variable resistive impedance presented to them by the *input terminals* of the volume controls. This, however, does not affect the frequency response of the system since a vacuum tube has the same flat (audio) frequency response regardless of the value of load resistance it faces. Any variation in the load resistance merely varies the gain rather than the response.

Although the plate load resistances are high (100,000 ohms), the resistance looking back into the output terminals of each pad is in the neighborhood of 50,000 ohms. Since the four are in parallel, transformer T-5 looks back into an equivalent impedance of about 50,000/4 = 12,500 ohms. This is the *apparent* source impedance for T-5 and is sufficiently low to enable the transformer to have a satisfactory low-frequency response.

The use of the 47,000-ohm resistor in series with each attenuator prevents the low plate resistance of the associated 1620 tube from connecting the left-hand terminal of the coupling condenser, such as C9, practically to ground and thus causing the condenser effectively to shunt the 100,000-ohm side of the pad. This could have an appreciable effect on the high-frequency response of the other channels in parallel with the pad in question; a similar consideration was discussed in connection with the electronic mixers (see Fig. 28 and discussion).

Referring now to Fig. 35, note that in the first stage of the OP-6 amplifier there is a ladder attenuator in the plate circuit to vary the gain to the next grid. Note further that feedback to the first grid is obtained from the top end of the ladder network. In this way, as the arm or swinger is moved down in the schematic to decrease the attenuation to the next stage, the attenuation in the feedback path is *increased*. Conversely, as the forward gain of the next stage is decreased, the feedback is increased. The value of thisfeature. is that in the case of a high-output microphone the amplifier not only requires less forward gain, but can also profitably employ more feedback to help cancel the stronger input signal and thus prevent the first stage from being overloaded.

In addition, a fixed amount of feedback is employed between the plate of the output tube and the cathode of the tube in the second stage. As a result of this, and also the choice of high-grade components, the amplifier is flat within 1 db from 30 to 15,000 c.p.s., and has a gain of 90 db, which is more than adequate to feed a telephone line from the microphones connected to the OP-7 pre-amplifier. A similar frequency-response characteristic is possessed by the OP-7 unit.

Finally, it may be noted in passing that the OP-6 unit can be equipped with a volume indicator, a monitoring jack permits the use of monitoring headphones, and by simply plugging in one of two cords, a.c. or battery power may be employed as desired.

SUMMARY

This concludes the technical assignment on speech input equipment. It dealt essentially with the audio components employed in the studio, and with their sequence in the system. First a block diagram of the general layout of the studio and control room was given, together with a description of studio procedure during rehearsal and during actual programs. In conjunction with this the functions of the master control room were analyzed so as to show the need for interlocking and pre-setting of circuits for multi-studio layouts.

A block diagram of the audio equipment then followed in order to present the series of operations that were performed on the initial microphone pickup, such as amplification, mixing, further amplification, monitoring, etc.

With this general survey completed, it was then possible to discuss the characteristics and design of the various components, particularly mixer circuits, since audio amplifier design had been discussed in previous technical assignments and required only a presentation of the overall characteristics such as gain, frequency response, etc. Finally remote pickup equipment was discussed, as this is a necessary adjunct of the normal studio equipment.

In the next technical assignment will be discussed the various control circuits that connect the above studio components together to form the complete terminal equipment.

AUDIO EQUIPMENT

EXAMINATION

1. (a) What is the difference in the function of the volume indicator and of the monitor loudspeaker in the control booth.

   (b) What is the function of the master control room?

   (c) What is pre-setting, and what is its value in program switching?

2. In the audio system, what is the function of each of the following:

   (a) The microphone pre-amplifier?

   (b) The mixer unit?

   (c) The program amplifier?

   (d) The line amplifier?

   (e) The isolation pad following the line amplifier?

3. (a) What two functions does the monitor amplifier have?

   (b) Explain how cueing may be performed.

   (c) What is a booster amplifier, and where is it used?

4. (a) What is the advantage of not loading the secondary of the input transformer of a microphone pre-amplifier?

   (b) Why should the microphone connections be color-coded?

5. (a) Give an example of the use of a mixer at the present time when high-grade, low-noise microphones are available.

   (b) What two fundamental requirements must a mixer meet?

   (c) What is the meaning of the insertion loss of a mixer?

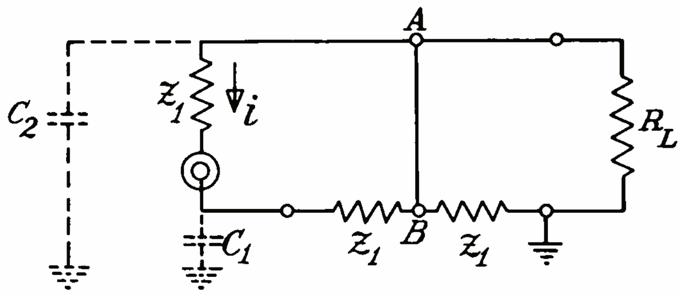
6. (a) Given six *balanced-to-ground* microphone inputs, each of 250 ohms impedance. A balanced output impedance is required. Design the proper mixer circuit to meet the above requirements. Show all components, including the pads to be used.

   (b) Calculate the insertion loss of this mixer.

6. (c) Show how by the use of isolation transformers of the correct impedance ratio, standard 250-ohm *tee pads* can be used throughout, even though the pre-amplifier inputs and the mixer output are still balanced to ground. Draw the circuit showing the values of all components.

7. (a) What is cross talk in a mixer circuit?

   (b) Given the following circuit. It shows a generator of internal impedance Z_1, feeding a pad of impedance Z_1. The pad is set for *infinite attenuation*, so that its shunt arm is a short circuit (line AB). The circuit is unbalanced-to-ground, but the pad is incorrectly connected, so that even though it is set for infinite atten-



uation, current does flow through the load R_L. Note that C_1 and C_2 are the stray capacities of the two sides of the generator to ground. Trace the cross talk current that manages to flow through R_L. (For convenience assume an instantaneous downward direction for the current through the generator, as shown.)

8. (a) What difficulty is encountered when an electronic mixer of several stations is desired, in which a common plate resistor is employed?

   (b) How is this difficulty obviated to some extent?

   (c) What two difficulties are encountered in using a supercontrol tube as a volume control?

9. (a) What is a bridging amplifier? Explain briefly.

   (b) An all-purpose amplifier is designed to act as a high-gain amplifier or as a bridging amplifier. Describe the difference in connections and operation.

9. (c) The above amplifier has an input impedance of 250 ohms as a high-gain amplifier, and 10,000 ohms as a bridging amplifier. As a bridging amplifier it has a gain of 20 db. What is its gain as a high-gain amplifier?

10. (a) In what way may a monitor amplifier differ from a program amplifier as regards frequency response?

(b) What are the fundamental requirements that must be met by remote pickup equipment?

(c) How is a remote program generally monitored? Why?