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ALTEC

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HANDBOOK

OF

ALTEC

SOUND SYSTEMS AND INSTALLATION DATA

CONTENTS:

Litho in USA CP-872-2K

1: TIME DELAY

A well-engineered sound reinforcing system is one which produces the reinforcement required of it without the audience being aware that a sound system is in use. This requires that the sound from the loudspeakers appear to emanate from the person talking. Speakers placed at either side of the proscenium destroy this illusion -- hence the audience is always aware of the sound system. The correct location for the speakers is in the vertical plane through the source of the original sound, either directly above, to the rear, or forward, of the person talking. If the loudspeakers are quite high, the sound will be clearly heard to come from above the 'talker' and the illusion again suffers. This problem can be solved in some cases by the use of a time delay device, making use of the Haas Effect.

When sound is projected from two separated sources with one source delayed 10 to 20 milliseconds behind the other, the undelayed source appears to be the only source -- even when the sound from the delayed source is up to 6 or 8 db louder. This is known as the Haas Effect. The writer witnessed a demonstration of the Haas Effect in which a small loudspeaker in a cigar-box size cabinet was set up in the center of the stage and a high quality loudspeaker, in a large cabinet, was set unobtrusively to one side. The sound to the large speaker was delayed and set louder than the sound to the small speaker. The wide-range sound, as heard, appeared to come from the small speaker.

The Haas Effect can be used in reinforcement systems to preserve directional illusion. The audience near the stage will hear both direct and reinforced sound, the latter from a loudspeaker above or to one side of the person talking. By introducing a time delay in the sound system, the sound may be made to seem to come from the 'talker.' The effect is limited to a level difference of 6 to 8 db so it will be effective only in the front portion of the seating area; however, to persons seated farther back, the angular displacement of the loudspeakers is not so great and the illusion may be quite satisfactory.

The time delay to be used must be such that the difference in arrival time of the direct and the delayed (reinforced) sound is in the neighborhood of 15 milliseconds; since the velocity of sound in air is one foot per millisecond, allowance must be made for the difference in the two path lengths involved. If the loudspeakers are 10 to 20 feet back of the microphone, the loudspeaker sound will be sufficiently delayed without the use of a delay device

Another use for a time delay device is in systems with distributed loudspeakers. The most useful type case is perhaps the situation where the rear portion of the audition area must be covered by loudspeakers located far ahead of the microphone; an example is an under-balcony area which requires a separate set of loudspeakers mounted from the rim of the balcony soffit. The acoustic time delay due to the difference in path length from the main speakers and the auxiliary speakers is so great that confusion results from the mixing of the two sounds. Adding artificial time delay to the system feeding the remote loudspeakers to correct the time difference removes this confusion. Making the added delay 10 or 15 milliseconds greater than the acoustic delay caused by the path length difference should have the effect of causing the separate speakers to be unnoticeable as a separate source of sound.

A fully distributed loudspeaker arrangement (a low-level system) should be considerably improved by artificial time delay, but the practicability of such an application is in some doubt, as the cost of the plurality of delay devices would probably rule them out.

Several time delay devices are on the market. Most of these employ endless magnetic tapes or a disc carrying a coating of magnetic dust, together with (usually) a single erase head and record head, plus one or more independent reproducing heads. The time delay results from the separation of the 'record' and 'playback' heads; however, very small time delays cannot be obtained with many of these devices (e.g, if the tape speed is 7-1/2 inches per second and the minimum possible spacing between heads is 1", the minimum delay obtainable is, therefore, 135 milliseconds. At 15 inches per second, the minimum delay would be 77 milliseconds). This may be the minimum delay limit possible for tape devices; the magnetic disc, however, can be driven at higher speeds and, hence, provide shorter delay periods, when required.

Some of the reverberation devices currently on the market are sometimes referred to as 'time delay' devices; many of these are not suited to the purpose described above.

1551A Jack Panel: The 1551A Jack Panel contributes great flexibility to a system; it mounts in only 1-3/4" of rack space (in a 19" rack or turret) and contains 12 pairs of jacks, in a single line, together with a designation strip for identifying each jack. The jacks are of the 3-terminal, 'normal thru' type, terminal 'A' connecting with the plug tip, when inserted, and terminal 'B' connecting through the springs and contacts when the plug is removed. The third terminal carries the shield to the body of the plug. If the normal source is connected to 'A' and the normal load to 'B', the plug will take over the source. If the load is connected to 'A' and the source to 'B', the plug takes over the load. If both source and load are connected to 'A', the plug will bridge the circuit without interrupting it.

1552A Meter Panel: In large systems which do not utilize components having built-in VU meters and associated range switches, the 1552A Meter Panel is provided. The panel occupies 3-1/2" of 19" rack or turret space and contains a standard VU meter, a range switch, range pads, and a terminal strip. The ranges provided will cause the 0 VU of the scale to represent +5, +10, and +15 VU on a 600Ω circuit. The range switch also includes an 'off' position to protect the meter from test signals which may overload it.

The VU meter is intended for measurement of program level; the program signal is a complex affair with frequent small peaks, less frequent high peaks, and occasional extreme peaks with low levels or dead intervals between. The time constant of the meter measuring this varying level has a controlling effect on the deflections obtained. The broadcast industry, some years ago, adopted a standard time constant, as well as a group of other standards defining the Standard VU meter. Any program level which produces extreme swings of the needle to 0 VU is said to be a level of 0 VU.

While the action of a standard VU Meter is fairly fast, it is not nearly fast enough to register the full value of the program peaks. The rms value of the peaks may be from 6 to 15 db above the maximum deflections of the meter needle. Sound equipment is normally designed to handle levels in dbm 10 db above the program level in VU; for example, a line amplifier intended to handle +8 VU, the maximum allowed to be sent over a telephone line, will be tested with steady signals at +18 dbm for distortion.

The VU meter is a 7500Ω AC voltmeter. When a steady AC voltage is applied, a deflection to the '0 VU' point on the scale (70% of full scale) is produced by the level in dbm corresponding to the switch position, provided the meter is connected to a 600Ω line. For any line, however, 0 VU deflection is produced by 1.38 volts with the switch set on +5 VU, 2.45 volts for +10, and 4.35 volts for +15. The modern Altec power amplifiers are driven to full output with an input of 0.9 to 1.0 volt, so the meter will not be deflected adequately when connected across the power amplifier input unless loss is set in the amplifier gain control. A loss of 14 db in the gain control will allow the meter to deflect to 0 VU at the +15 VU setting when a steady signal is applied sufficient to drive the Altec 1568A and 1569A amplifiers to full power.

When the meter circuit is connected across higher voltages than those corresponding to +5, +10, and +15 dbm in a 600Ω circuit, it is necessary to add a suitable pad ahead of it. Terminals are provided in the 1552A for mounting the elements of the pad, which is a two-element type having the series arm toward the source and the shunt arm across the output feeding the normal circuit of the panel. A handy point at which to connect the Meter Panel is the 70-volt system output.

To use the Meter Panel across a 70-volt output, an L-pad must be added to reduce the voltage to the meter panel circuit without changing the input resistance to the combination. This will consist of a series arm R1 toward the source, and a shunt arm R2 across the meter panel input circuit, with values as given in the following table. These are for various possible choices of meter deflection at full output (70 volts) steady signal. For program signal, the deflections will be about 10 db or 10 VU less.

TABLE I

Scale Setting at which 70v will produce deflection of 0 VU	Corresponding Input Volts for 0 VU deflection on +5 VU Scale	Pad Values R1	Pad Values R2	Pad Correction
+5 VU	70v	7350	151	34 db
+10 VU	40v	7230	272	29 db
+15 VU	22v	7072	496	24 db
+20 VU*	12.5v	6700	930	19 db
+25 VU*	7v	6000	1830	14 db

* Scales not provided in the 1552A but cases included here to provide pads of less loss for amplifiers not fully driven.

To read program level from the Meter Panel equipped with a supplementary pad, add the following three corrections to the scale reading of the meter:

(a): The scale correction (+5, +10, or +15 VU)
(b): The pad correction from above table
(c): The impedance correction, as follows:

Load Impedance	Impedance Correction
600Ω	0 db
150Ω	+6 db
124Ω (70v, 40 watts)	+7 db
62Ω (70v, 80 watts)	+10 db
32Ω (70v, 165 watts)	+13 db
16Ω	+16 db
8Ω	+19 db
4Ω	+22 db

1553A Monitor Panel: The panel occupies 8-3/4" of 19" rack or turret space and contains a 401A speaker, a 15064 70-volt transformer, an adjustable L-pad volume control and a terminal block. The Altec 356A Amplifier may be used when a monitor amplifier is required.

1554A Power Distribution Panel: The panel occupies 5-1/4" of 19" rack or turret space and contains a 20-ampere toggle switch, pilot light, two AC outlets on the front and six on the rear. The rear outlets and pilot light are energized only when the toggle switch is in the 'on' position, while the two outlets on the front of the panel, which are intended for general purpose usage such as powering auxiliary lights, clock, soldering iron, etc., are not controlled by the switch. All outlets are of the 3-pin type.

1555A Switching Panel: The panel occupies 3-1/2" of rack space and contains 12 independent, 2-circuit, 3-position lever switches. A designation strip extends across the top of the panel to provide a means of writing in the identification of each switch; three write-in cards are provided at the end of the panel to identify the three positions of the switches. The panel is primarily intended for speaker zone switching, the three positions allowing, for example, music, paging, and 'off' to any speaker or speaker zone; the panel may also be used for microphone switching.

1557A Speaker Selector Panel: The 1557A Speaker Selector Panel is designed for use in sound distribution systems for schools, factories, offices, and all other types of installations requiring that each remote speaker station be switched to any one of four program lines from a central point. The 1557A contains a bank of twenty-five, 4-position lever switches. Directly above each switch is a lucite-covered removable designation strip, on which the function of each switch may be written; 25 5/16" holes are drilled behind the strip so that indicator lights may be installed by the contractor, if desired. (Indicator lights and necessary hardware are not furnished).

When used with the Altec 1556A 'Altalk' amplifier, 1558A Program Selector Panel (below), and 1559A Emergency 'All Call' Panel (also following), the 1557A permits instantaneous selection of each of the four program sources for distribution to each of twenty-five stations. The 1557A is completely wired internally; terminal connections (screw and/or wire wrap) are provided for two program inputs (A & B), 'IC' (for 2-way intercommunications) and 'off' (for 'all call' or 'all speaker') on a select basis, together with 'all call' for paging or emergency announcements. The 1557A measures 3-1/2" high and mounts in a standard 19" rack or turret; the panel is finished in Altec Green.

1558A Program Selector Panel: The Program Selector Panel contains two, 10-position rotary switches, each of which is individually identified as Selector 'A' and 'B', respectively, used to select any of 10 program sources for it's respective program channel. The 1558A is supplied completely wired with a terminal board on the rear; each terminal pair is numbered and a lug is provided for terminating the shield of that pair. All terminals of the panel are wire wrap and/or screw type, providing the greatest flexibility of installation in a minimum amoun' of time. The panel measures 3-1/2" high and mounts in a standard 19" rack or turret; the finish is Altec Green.

1559A 'All Call' Emergency Panel: The 1559A Emergency Panel is designed for school and paging systems and may be used in conjunction with the Altec 1556A 'Altalk' amplifier, 1557A Speaker Selector Panel, and 1558A Program Selector Panel -- the basic components for all intercommunication systems. The 1559A provides switching facilities for both 'all call' and 'all speaker' functions. Complete flexibility for 2-way intercom systems is obtainable, due to the unique strapping arrangement available on the rear terminal board.

The 1559A provides many switching functions, among them the ability of switching the 'all call' or 'all speaker' bus and feeding into four speaker distribution channels (selectable); facilities are provided to permit the use of a separate 'stand by' power amplifier or a program channel amplifier for the 'all call' facility. Also on the rear terminal board are three spare terminals for 'normal through,' or switching (SPDT) of control relays, lights, or other desired DC functions. The mode of operation is clearly indicated through a 'window' located above the switch control knob; when installed, the desired function ('Normal IC,' 'All Call', or 'Normal IC', 'All Speakers') may be chosen by rotating the white mode identification disc. The switch is not a spring return device, hence, when the desired function is selected, it will remain in that position -- eliminating the need for holding and thereby allowing 'hands free' operation.

The 1559A panel measures 3-1/2" high and mounts in a standard 19" rack or turret; finish is Altec Green.

III: PRIORITY SYSTEMS

Priority provisions are specified for a number of industrial systems, whether or not the industry itself is engaged in defense activity. In some cases, a number of priority levels must be provided. The following is typical of the priority system requirements of large aircraft plants:

1: Area Page -- no priority
2: Area Boss -- capable of taking over from all other microphones in his area.
3: General Page -- capable of taking over from all systems, in all areas, interrupting local paging that may be in progress.
4: Big Boss -- capable of taking entire system from General Page originating stations.
5: Tone Signals -- generally required to interrupt any announcement of lower priority.
6: Security Officer -- top priority over entire system.

(Note: In some instances -- such as the system described in the final portion of this section -- the respective priorities of 5 and 6, above, may be reversed.)

Priority requirements extend downward from the foregoing 6-level example to the relatively simple case of only one degree of priority ('over-call') without signal lights, the most common example of which is used to suppress background music while paging. Compressors afford a handy means of obtaining a 2-level priority system such as this without complication of special cables, ganged switches and/or relays. When a compressor is used in this type of 2-level priority system, the gain of the priority microphone preamplifier is set at a higher level than the gain of the non-priority microphone preamp (or the background music preamp); by this means, the priority mic signal will cause the compression to increase and, thus, substantially suppress the non-priority signal which may be transmitted at the same time. If, for example, the compression settings are such that the non-priority signal produces 10 db of compression and the priority signal level to the compressor is 20 db higher than the non-priority input, the compression will increase to 25 db on the priority call. The priority signal, therefore, will be reproduced 5 db louder than the non-priority signal which becomes virtually inaudible (15 db below 'normal' level and 20 db below the priority call).

If a greater difference between the reproduced level of priority and non-priority signals is required (e. g., 30 db), the difference in input levels to the compressor is made 30 db and the compression for the non-priority signal reduced to about 5 db in order to avoid overloading the compressor when a priority call is made. The non-priority call then reproduces 20 db below normal level and the priority signal is about 10 db louder than the non-priority call (or background music signal). Care must be taken to assure that the system amplifier(s) is(are) not overloaded by this level increase.

The operating point referred to in paragraph 2 leaves a good compression range available for the other purpose of the compressor -- that of equalizing variations in voice/music levels. The settings referred to in paragraph 3 reduces the range of the compressor by a considerable amount but does not eliminate the range entirely.

In larger systems, priority is usually accomplished by microphone switches or by relays controlled by the microphone push-to-talk switch (as described in the system detailed at the conclusion of this section). In Figure 1, all switching is done by the 7A Microphone Switch Kit in the 26A Desk Stand. In this type of connection, the microphones are all in series, each being normally shorted out by it's own switch: When the switch for #2 is actuated, the short is removed from #2 and the line to #3 is shorted; thus Mic #2 is given priority over #3 and, similarly, #1 has priority over #2 and #3. This system can be extended to many microphones on a given priority level and to any number of priority levels.

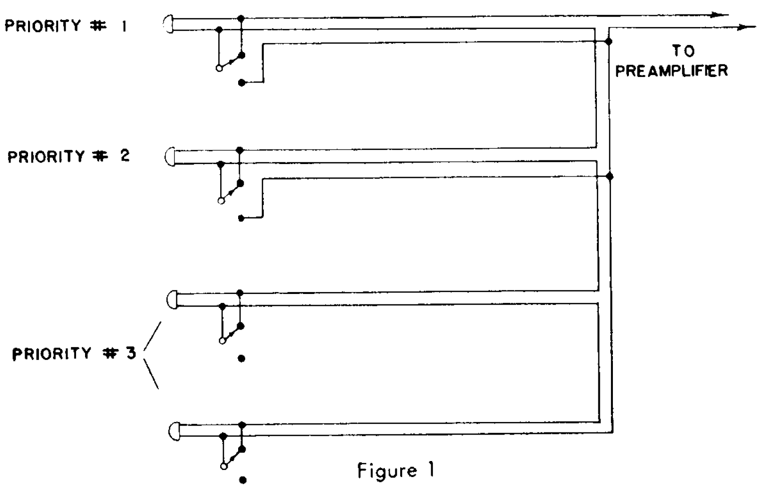


Figure 1

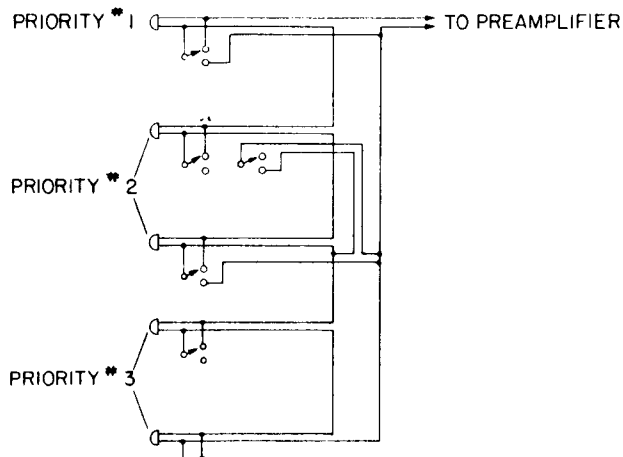


Figure 1A

Figure 1A shows connections for the use of more than one microphone on a given priority level: In this system, 3-conductor shielded cable must be used for each priority microphone. It may be preferred to use control wiring and relays in place of the third signal conductor, the relays being allowed to perform their switching in the low-level lines when the system layout makes no other location available. (The problem of connecting low-level circuits to relays is discussed under the heading Sound System Wiring.) It is safer to keep relays out of the low-level circuits; therefore it is often preferred to perform switching after the preamplifiers when this is possible. If each preamplifier employs a build-out resistor in series with it's output, the relay may be arranged to short out the circuit just ahead of this resistor.

Figure 2 illustrates circuitry for three microphones, with three levels of priority, controlled by relays.

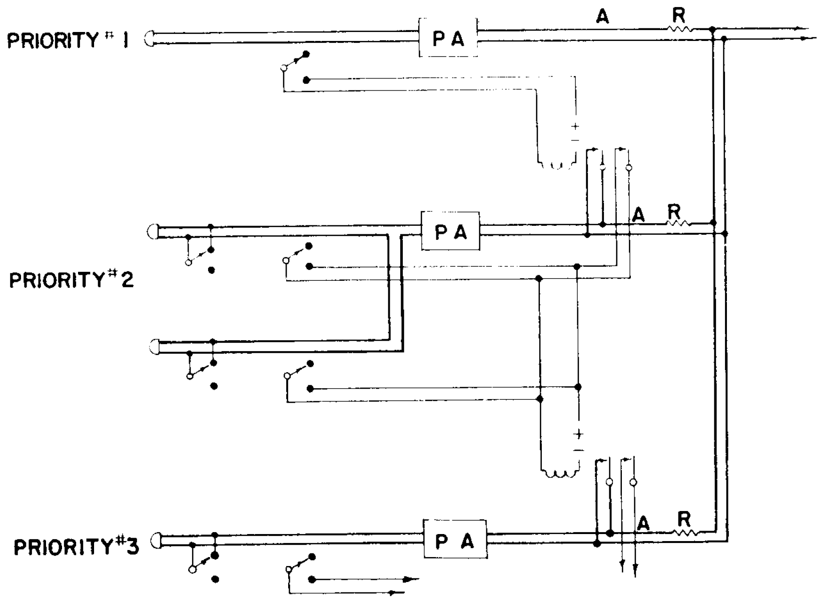


Figure 2

IV: COMPRESSORS

The compressor does automatically what an experienced control operator tries to do -- namely, it lets a rising voice or orchestra level reproduce louder, but not as much so as the original sound and it allows the weaker voice/orchestra level to sound weaker (as it should) but not let it get lost from audibility. At the same time, the compressor protects the amplifier, speakers, and the ears of the audience from extreme volumes. Altec compressors are provided with a range of attack- and release-time characteristics that meet virtually all sound system needs. Attack is fast -- 1/20 of a second -- much faster than a human operator can respond, but not so fast that 'thumps' are reproduced by the control action in the amplifier. Release is much slower and adjustable between 0.3 and 1.3 seconds, in order that voice inflections may be permitted to carry over, and in order to avoid wide changes of gain during brief pauses.

Practically every PA system can be improved with a compressor amplifier; the benefits easily justify the cost:

1: <u>All systems that are unattended</u>: Even if only one person originates all paging calls, the chances are '10 to 1' that his or her voice level and mic distance will vary, due to the fact that he or she invariably has other duties. When calls originate from various persons, the compressor becomes vital.

2: <u>Systems that are attended only part of the time</u>: For large-scale events, an operator is employed; for lesser functions, the system is merely turned on and left unattended. The compressor greatly increases the number of functions for which the operator will not be required and, naturally, improves the operation of all unattended situations.

3: <u>Operation with attendant at the mixer controls</u>: In these cases, the compressor corrects for human frailties, response of control is speeded up and errors of judgment are minimized. An example is the use of compressors for television opera pickup, in which a powerful voice may sound the first note of a selection at high volume. A mixer operator cannot respond fast enough and, in trying to anticipate the great burst of tone (rehearsal is never at full voice), he either under-compensates or overcompensates. The Altec 436C is as fast as necessary, and it's general quality of audio performance and reliability adapt it for use with associated equipment which must meet the most exacting broadcast requirements. The need for a relief operator is usually eliminated.

4: <u>Priority voice-over-music or voice-over-voice</u>: In systems which distribute music between paging calls, the compressor can be made to fade the music automatically the moment the announcement begins. Non-priority announcements may be similarly faded when an announcement is made from a priority source (cf: Section III, <u>Priority Systems</u>).

The adjustment of level to compressors involves a few special considerations: It has been complained that compressors increase the tendency for system feedback. This is untrue; the idea originates from experience when system gain is set up for strong voices. The feedback problem is always a weak-voice problem and the compressor does not alter this basic fact. For the weakest voice, the level to the compressor must produce no compression and the system gain, reinforcement, etc., must be limited to a suitable margin below the feedback point. The strong voices then produce much stronger levels at the compressor input but only moderately stronger loudspeaker outputs. The compressor does not aid or hinder solution of the feedback problem; if feedback does occur, the compressor will prevent it from reaching a level higher than that of a weak voice unless adjustments are grossly incorrect.

It is not easy to determine in advance the exact program level that should be supplied to a compressor to produce a given amount of compression, for the reason that the compression depends on a kind of average of the instantaneously varying magnitude of the program signal. For this reason, the compressor is given a wide operating range to eliminate the need for exact settings, and it is provided with a meter indicating the extent of compression directly.

The meter is an important aid in most uses of the compressor. It indicates, at one extreme, when the input level from weakest voices is just reaching the threshold, so that compression will become effective on all voices from that level upward; at the opposite extreme, the compressor indicates when the levels are excessive and approaching overload.

The levels at the compressor must be selected on the basis of the factors discussed. This requires, in most cases, adjustment of gain between the compressor and power amplifier, a requirement met by the gain controls of the power amplifier. In the case of the Altec 260A amplifier, a gain control must be added.

The master gain control preferably should be located after the compressor, in order that the chosen operating point will not be disturbed by master adjustments.

V: IMPEDANCE MATCHING

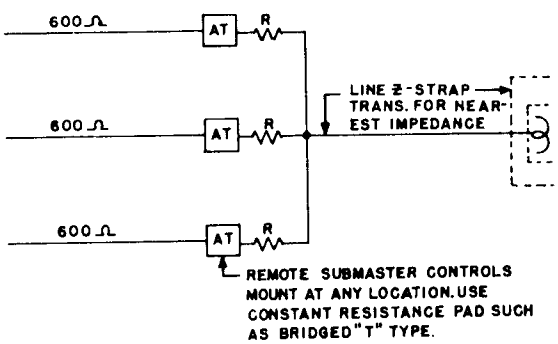
Impedance matching is a sweeping term, meaning different things in different applications. In sound systems of the 'old school', equipment was usually combined on the basis of equal impedance. This is now almost the exception; the term generally means choice of impedance.

Input of a Preamplifier: Low-impedance inputs of amplifiers using vacuum tubes are invariably equipped with step-up transformers. Sometimes the grid side is terminated with a resistor such that a 30:70,000Ω transformer, terminated with 70,000Ω, actually presents 30Ω at the input. In other designs, the transformer may work into the grid without a terminating resistor, in which case, the actual impedance seen at the 30Ω input terminals would be (perhaps) 10 or more times 30Ω through the mid-range; perhaps 60Ω inductive at the low-frequency limit, and capacitive or inductive in the same order of magnitude at the high-frequency limit. If a source of 30Ω is connected, these impedances will provide a response which is flat throughout the midrange, and down about 1 db at the rated frequency limits; thus, there may not be equality of impedance between microphone and amplifier, but no problem is presented to the user -- since the proper impedance for the source is stated by the manufacturer. If a 150Ω microphone is used at the 30Ω input of an unterminated transformer, the mid-range gain will be roughly 5 db higher, but the response will drop off by possibly more than that amount at both frequency limits. Thus, correct impedance selection at an unterminated input provides the maximum gain with the rated frequency response and, in this case, impedance of the source is only 'matched' to the impedance rating of the input.

Input of a Power Amplifier: At the input of a power amplifier, there may be a 70k resistance and no transformer. Does this mean that a transformer must be provided, in order that the amplifier may be fed from 70k ohms? -- Decidedly not: If the voltage sensitivity of the amplifier is 1 volt, then 1 volt at the input terminals will produce full output, regardless of the impedance of the source that delivers the volt. No impedance or frequency response consideration applies importantly at this point. If the source, for example, can supply only two-tenths of a volt, then a 600-to-15,000Ω transformer (such as the Altec 15095) will raise the voltage to 1 volt (the voltage step-up is the square root of the impedance ratio of the transformer; in this case, the square root of 25, or 5. If the transformer is used, it becomes necessary to have a source impedance which agrees with the nominal input impedance of the transformer taps, if the proper response is to be obtained. A more common reason for using the transformer in the input of a power amplifier is to isolate the circuit from ground and permit the use of a balanced line to the amplifier.

Output of a Preamplifier: The Altec 1566A Preamplifier uses a cathode follower as an output stage. The output impedance (i. e., impedance looking back into the amplifier from the output) is in the order of 1,000Ω, but to deliver rated output without significant distortion, the tube must work into a high-impedance load of, perhaps, 15,000Ω or more. Outputs of preamplifiers are usually paralleled at a mixing bus; if only two are paralleled, #1 will be loaded with the 1,000Ω output impedance of #2; if 6 are paralleled, each preamplifier will be loaded with 160Ω, greatly reducing it's power capacity and leading to distortion. This is prevented by building out each output with a series resistor of 15,000Ω so that each output tube works into at least 15,000Ω. Of course, there is a loss in such a network of build-out resistors -- namely 6 db for 2 channels, 10 db for 3, 12 db for 4, 14 db for 5, 15.5 db for 6, 18 db for 8, and 20 db for 10 channels. If there is an equal resistance (e. g., 15,000Ω) shunting the combination by the circuit following, this may be counted as an additional channel when figuring the loss. When the mixing point immediately follows the mixer gain controls (as in the Altec 342B amplifier), an additional purpose of the isolating build-out resistors is to avoid having the low impedance of the mixing point upset the logarithmic taper of the mixing potentiometers.

Combining Attenuators: When several channels lead to individual mixer attenuators of the 'Bridged-T' or ladder types, and the circuits are to be combined after the mixers, it is desirable to have the impedance loading each attenuator equal to the nominal impedance of the attenuator. This is one of the examples of matching for equality; it's purpose it to maintain the circuit impedance at a design value, regardless of the attenuator step. The method is shown in Figure 4:



NO. OF LINES	2	3	4
MIX LOSS-DB	4.8	7	8.4
R (OHMS)	300	400	450
INPUT TRANS., IMPEDANCE	600	300	300

Figure 4

The question often arises whether a stepped attenuator should be terminated with a resistor when the following equipment has a much higher impedance. It is a little-known theorem of the 'Variable-T' or 'Bridged-T' attenuator that the steps of loss are not changed when the impedance of the source only, or of the load only, is changed from the normal value -- so long as the impedance looking in the other direction is the value for which the unit is designed. In other words, a 600Ω, 2-db-per-step 'Bridged 'T' attenuator remains exactly 2 db per step if, for example, the load impedance is made 100Ω, so long as it continues to work from a 600Ω source. The same is true if the source is changed, but the load remains 600Ω. However, in the case of the 100Ω load, the actual input impedance decreases as the loss of the unit is reduced, becoming 100Ω at the zero-loss step. Similarly, if the load is removed, the output voltage rises 6 db, regardless of the step; the impedance looking back toward the 600Ω source remains 600Ω at all steps, but the input impedance rises at the low-loss steps and becomes infinity at zero loss. With this in mind, it is often unnecessary to terminate an attenuator in a matching resistance.

The ladder attenuator is not as ideal in this respect as the 'T', due to the reduction of it's output impedance to about 2/3 on the last steps toward zero loss. Removal of the nominal load from the ladder causes a 6 db rise in output voltage, except on these last 6 steps or so. On the last step, the rise is 4.4 db, and the change -- as the control is advanced to the last step -- is 1.2 db instead of the nominal 2 db. However, unless these principles are well understood, it is usually the safest practice, generally, to terminate equipment in it's rated load.

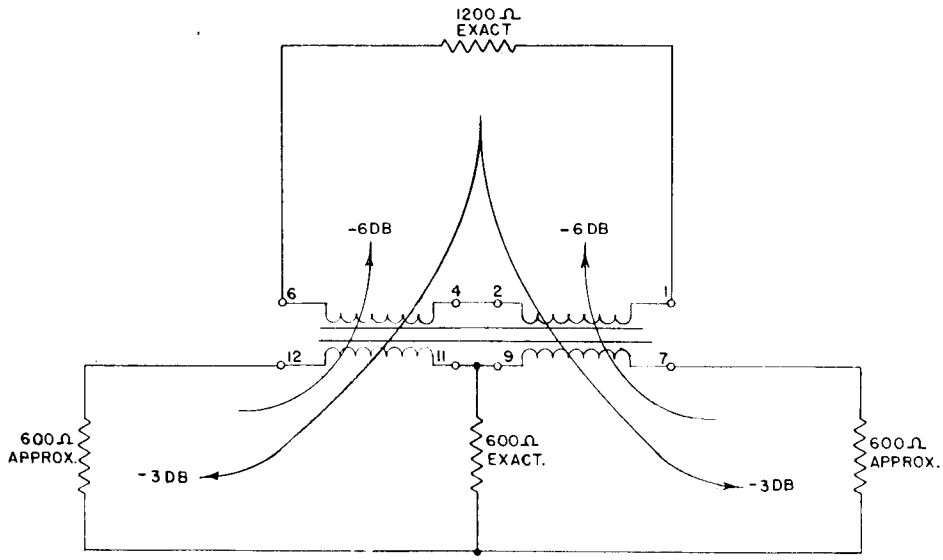
Mixing at the Output of Power Amplifiers: Suppose 40 watts of power is required of a system having 8 inputs: Why not use two 342B amplifiers and parallel the outputs? The internal impedance of the 342B is 20% or less of the nominal load impedance, or, for example, 3Ω for the 16Ω output. Amplifier #2 then sees a 16Ω speaker load in parallel with the 3Ω output impedance of amplifier #1, or a combined load impedance of 2.5Ω. The amplifier will deliver only a few watts at low distortion into such a low load impedance. This situation differs from the case of amplifiers of equal gain, paralleled at both input and output, as in the latter equal voltages in phase exist at the output terminals of both amplifiers before their outputs are paralleled. On connecting their inputs together, no current from one can flow into the other because their voltages oppose. For the case of mixing different signals, the voltages are not related to each other and the effect is the same as though one has no signal. If build-out resistors are used, as with preamplifiers, their value would be 13Ω and the output shifted to the 8Ω tap; then the power available for the load would be about 6 db less than 20 watts, or 5 watts.

Note, however, that power amplifiers with inputs and outputs paralleled will perform well only when they are in perfect balance. Since perfect balance cannot be reliably maintained and, since performance deteriorates rapidly as unbalance increases, such paralleling is not recommended. The Altec 7740 'SEQUR' panel provides the ideal means for safely paralleling amplifiers.

Fixed Pads: Pads are often required in sound systems. There is nothing mysterious about fixed pads; they can be made up quite easily. The values for the resistors can be readily worked out with the help of the guides given in Table 2. Values are tabulated for many loss values for pads of 1:1, 2:1, 4:1, and 10:1 impedance ratios, based on 1000Ω, and these are easily scaled up or down for any impedance. Simple formulas are also given for computing any pad. The computed values will usually come out odd, but the nearest RETMA 5% values are easily accurate enough for practically any purpose. Half-watt elements are large enough for circuits carrying signal levels up to 20 dbm.

Commonly-used pads are either of the 'T' (or 'H') type or the 'L' type. To obtain a stated loss and a stated impedance ratio, the 'T' (or 'H') is used, provided the combination of loss and impedance ratio is possible of attainment. For example, a loss of less than 8 db cannot be obtained with a pad having an impedance ratio of 2 to 1; this calibration is, therefore, an 'impossible calibration.' If the purpose of the pad is to provide an impedance ratio at the minimum loss, the 'L' pad is used, the series arm extending in the direction of the higher impedance. If this type of pad is to be used in a balanced circuit, the series arm is divided into two equal parts, one being placed in each side of the circuit.

The hybrid transformer has a very interesting and, on occasion, useful property; Figure 5 illustrates the way it works: If the signal originates from the lower left, transmission will occur to the upper load and none of this signal will appear inthe lower right termination. Conversely, if the signal originates in the lower right, it will again be transmitted to the upper load, while no part of it will appear in the lower left. These properties require that the balancing resistor connected to the center point of the transformer winding be set to the proper value, relative to the upper termination.



15036 TRANSFORMER AS HYBRID

Figure 5

To understand this operation, consider the signal to originate from the lower left, and assume the impedance looking into 11-12 to be exactly equal to the value of the balancing resistor. Then the voltage drop from 12 to 11 will be duplicated exactly across 9 to 7, which would seem to induce a voltage in the circuit of the right-hand termination. However, this voltage is exactly equal -- but opposite -- to the drop across the balancing resistor, so that no voltage is available to cause current flow through the right-hand termination. Since one-half the voltage is lost in the resistor, the signal originates in the upper termination, the two loads may be considered in series with each other and no voltage will, therefore, appear across the resistor. There is no loss in this resistor and no loss in transmission. Due to equal division between loads, each will receive 3 db less than the original signal power.

The Altec 15036 transformer makes an excellent hybrid coil for use at the usual line impedances; the Altec 15067 transformer, as used in the 7740 'SEQUR' panel, is an important application of the hybrid coil.

Low-level signals, such as microphone outputs, are always carried in single-circuit or two-conductor shielded cables. In either case, it is essential that the shield be insulated from contact with building grounds by an outer jacket. For low-impedance microphone lines, the cable should have two conductors, well twisted within the shield, the shield being grounded at the amplifier. For high-impedance microphones, the single conductor shielded cable is usually preferred, as it generally has lower capacity for a given cost and slight pickup is harmless in the return conductor (shield) over the relatively short lines that can be run at high impedance.

Long, 2-conductor lines are generally quieter if the circuit is not grounded at one side, but, in troublesome cases, it is worthwhile to try them one-side grounded, center tap grounded, and ungrounded. If no center tap is provided, two exactly-equal resistors, having a value double the circuit impedance, connected in series across the line at the amplifier input, provide a mid-point for grounding. The 30/50Ω tap of the Altec 4722 plug-in input transformer is an accurate mid tap for the 120/200Ω terminals, and the 150Ω tap on the Altec 15095 plug-in line transformer is mid tap for the 600Ω terminals.

A discussion of the several means by which noise is introduced into the system by way of the wiring might be helpful: Noise pickup may be by electromagnetic, electrostatic, or conductive means.

A Magnetic Pickup: Magnetic Pickup results when an AC magnetic field links the circuit; the two sides of the circuit and the source and load constitute a loop.

In Figure 6, A and B are circuit conductors. Line #1 links no conductors and is harmless; line #2 links 'A' only, inducing a hum voltage in 'A', causing current to flow through the circuit and produce a voltage across the load (such as an input transformer), thus introducing noise into the system. Line #3 links both 'A' and 'B', producing equal and opposite currents which cancel.

Two methods are available to minimize magnetic pickup: If the two sides of the line are extremely close together, very little flux can link one side without linking the other as well. If the conductors are well twisted, for any flux that links one side, there will be flux to link the opposite side an inch or two farther along the cable -- the resulting induced voltages on the two sides of the line cancelling each other out. Both of these measures are highly effective in a good microphone cable. The shield has practically no effect on magnetic pickup.

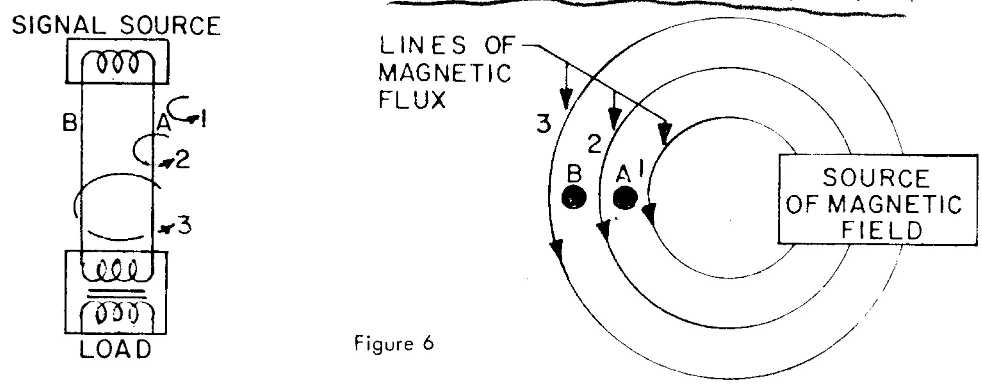


Figure 6

B Electrostatic Pickup: Electrostatic pickup is due to capacitance (usually very small) between a circuit conductor and some other conductor in the vicinity at an AC potential from ground. Electrostatic fields exist practically everywhere in the unshielded spaces in modern buildings and, often, outdoors. Electrostatic pickup by any conductor is completely prevented by a perfect grounded shield.

C Longitudinals: Cable shields are usually good enough for most purposes, but they are not perfect and, under some conditions, a small voltage is picked up electrostatically. In the case of 2-conductor cables, the potential of both sides of the line is raised and lowered by this pickup. At the input transformer, both terminals are raised and lowered by this voltage together and in phase and, in this case, the noise voltage is between the far end and the near end of the line, giving rise to what are called 'longitudinal' currents.' If one side is grounded, the voltage to ground exists at only one terminal of the transformer; hence, there is a voltage across the winding and a noise voltage is transmitted. However, if both sides are equally free of ground, no voltage exists across the winding and no noise voltage is transmitted. This explains the advantage of the ungrounded input circuit. If the capacity to ground of one end of the transformer winding is not equal to the corresponding capacity of the other end, an unbalance exists and some part of the noise voltage will be transmitted. The use of a pair of equal resistors across the line, with the mid-point grounded, balances the line even if the transformer is not balanced, because capacities in the transformer are small and their impedances are so much greater than that of the resistors that the latter take control. It may, in some instances, be more effective to ground the exact mid-point of the winding. In long telephone lines that may run parallel to power lines, longitudinal currents are sometimes considerably greater than the signal currents and it is necessary to use a transformer at the receiving end having extremely accurate balance of capacities of the windings. The Altec 15036 repeat coil, an outstanding example of such a transformer, attenuates longitudinals by 80 db.

Techniques for Minimizing Noise Pickup: The cable termination at the amplifier is extremely vulnerable to pickup, since the conductors are not twisted for a short space. They may not be maintained close together and they may be exposed to an AC electrostatic field due to proximity of a conductor at AC potential without an intervening shield. (Altec amplifiers use magnetically-shielded power transformers to minimize radiated magnetic field.) The proper technique is to:

1: Carry the shield as close as possible to the terminals
2: Reduce the untwisted length of conductors to a minimum
3: Separate the wires as little as design of the equipment makes possible.

These basic rules must never give way to a 'pretty arrangement' with the shield well stripped back and the conductors nicely separated and neatly arranged with the wide, sharp bends.

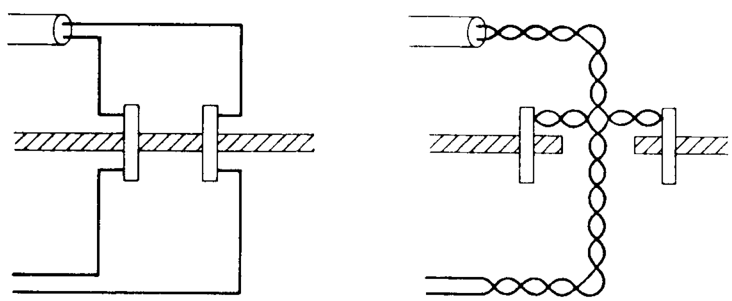
It is easy to observe the rules when microphone lines are brought up to an amplifier as the design of the equipment encourages use of the proper technique. When low level lines are brought to relays or other equipment ahead of the amplifier, great care and thought must be exercised. The operating coil of a relay radiates a power magnetic field which often has a fast rise, and the coil terminals may radiate a sharp electrostatic field. It is unavoidable to separate the leads to reach the contact terminals and the circuit is carried over the spring leaves which present an open loop. In these regions, the circuit is often unshielded, always untwisted, and separated more than desirable. Few relays are well suited for use in low-level circuits; however, relays are successfully used in some cases. If they must be used, select a relay with precious metal contacts, signal terminals removed from the operating circuit terminals, and short contact springs spaced close and parallel to the swinger springs. (Assurance of good contact is enormously improved with bifurcated springs having double contacts.)

Avoid use of additional terminals, the cables being brought directly to the relay terminals. If terminals are required for joining lines, make all connections on the same side of the panel, and place the terminals as close together as practicable.

In Figure 7, 'Dangerous' illustrates a common error in wiring to terminals extending through a panel. If any magnetic field is present and the signal level is low, pickup is practically sure to result. 'Safe' shows how connections can be handled so that the magnetic pickup at this point is reduced at least 60 db.

If interwiring of the signal lines between relays is necessary, don't loop one side of the circuit from one relay to the other without carrying the return along, the two sides being well twisted between relays. If one side of the line must be detoured, there will always be at least two wires involved (even though they are the same side of the line) and these must be twisted together. If the relay transfers between circuits, there may be three wires involved; these should be twisted together.

In running down the source of noise pickup, it is often helpful to determine whether it is due to electromagnetic or electrostatic coupling. It is usually an easy matter to localize the portion of the system where the noise originates. If it disappears when the circuit is opened at the source, it is probably due to electromagnetic coupling, since the line must be able to carry current for the voltage induced in a conductor to be developed across the load; also, shorting the source end of the line usually increases the noise several decibels. If the noise remains, or becomes greater when the source end is opened, it probably enters the circuit by electrostatic coupling. Aural study of the noise with a good speaker is very helpful: electrostatic pickup of power line frequencies is usually rich in harmonics; electromagnetic pickup is strong in fundamental -- 60 or 120 cps, depending upon the source.



DANGEROUS Figure 7 **SAFE**

Electrostatic pickup, as previously stated, is eliminated by shielding. A convenient method of experimenting with shielding in close quarters is to use a piece of metal foil with paper glued to both sides, the shield being grounded by any handy means -- such as by the use of a clip lead.

It is not possible to do more than generalize in discussing this subject. The rules given above must be checked over in every case. A given layout must be carefully studied under each condition of the circuit relay or switch positions, noting the signal currents in each conductor and insuring that the conductor carrying the same current in the opposite direction is never separated 1/4 inch and twisted if possible. Electrostatic pickup can always be prevented by the use of proper, grounded, shielding elements.

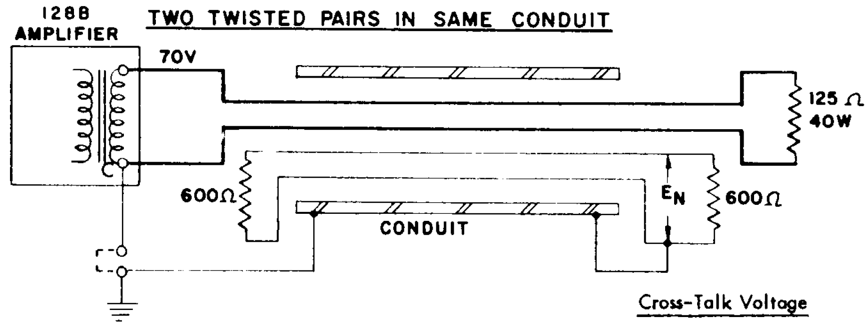
<u>Grounding, Balancing, Shielding -- Effects on Crosstalk:</u> When a plurality of lines, carrying different programs or signals, are run together in the same conduit, they tend to induce crosstalk currents into each other. When planning such a run, answers are needed for several questions:

1: Must the cost be paid for shielded cables, or will twisted pairs suffice?
2: Should transformers be used?
3: Should the circuit be grounded?
4: What about capacity?
5: What about line loss, selection of conductor size?

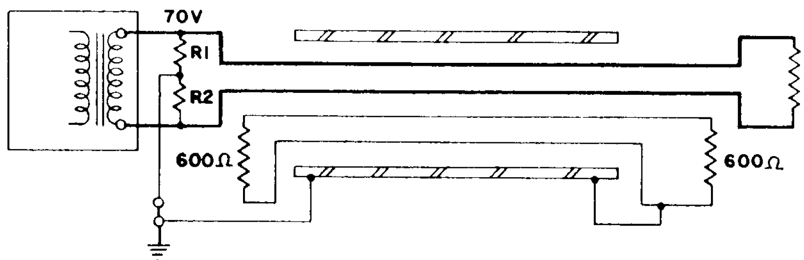
Crosstalk is induced (a) electromagnetically, due to unbalanced coupling between one circuit and others; (b) electrostatically, due to unbalanced capacitance to other circuits, or to the conduit if it carries current and thus developes a voltage difference between one and the others, or to it's own or other shields carrying current.

Two wires of a pair <u>must</u> be twisted; this insures close spacing and aids in cancelling pickup by transposition. In the measurements reported in Figure 10, all pickup was capacitive. The twisting of the leads apparently effectively eliminated inductive coupling.

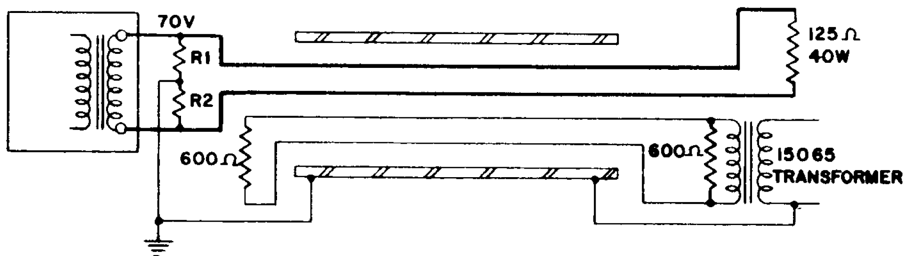
Measurements were made on a twisted pair run in the same conduit with a similar twisted pair, the latter carrying signals at 70 volts. The length measured was 250 feet, but check measurements made for half this length produced half the voltages; therefore, the results to be expected with longer runs are increased in proportion in the following tabulation. The disturbing line was driven from the 70-volt terminals of an Altec 128B amplifier and the line was loaded at the far end with 125 ohms, thus transmitting 40 watts. The figures are for 1,000 cps, but the voltages measured at 100 and 10,000 cps were one-tenth and ten times these figures, respectively. The noise values measured were induced by capacitive coupling:



	Cross-Talk Voltage		
	250'	1000'	2000'
1: Amplifier common grounded	0.1v	0.4v	2.0v
2: Ground removed, 70-volt line floating	0.014v	0.06v	0.3v



3: 70-volt circuit grounded using pair of resistors matched to 10%	0.005v	0.02v	0.1v
4: Same, with resistors matched to 1%	0.0006v	0.0025v	0.012v



5: 70-volt circuit same as in (4), above	0.0004v	0.0016v	0.008v
	(Referred to 600Ω side of 15065 trans)		
6: Same conditions as (5), except that disturbed line is 2-conductor twisted shielded cable	0.0002v	0.0008v (or less)	0.004

Figure 8

The above voltages may be converted to crosstalk ratios, relative to 70 volts as follows:

Voltage	db Below 70v
.00007	100
.007	80
.07	60
.7	40

Objectionable mid-range crosstalk ratios may be considered to be between 30 and 60 db, depending on listening conditions, noise, quality standards, etc. Therefore, some of the cases listed are unsatisfactory, while others represent satisfactory conditions for lines up to 2 miles or more in length.

Capacitance of Line:

The capacitance of a line loads the amplifier, reduces it's power capacity, and increases it's distortion. It will cause some loss of high frequencies in the case of line-level circuits that are quite long, but this effect is negligible in the case of power amplifiers. The table Length of Cable for 3 db Loss At 10,000 cps gives some examples that will permit ready evaluation of this effect. For 70-volt power circuits, a good rule is to avoid capacitance that exceeds, in microfarads, the value:

$$\text{Max } C = \frac{W}{400} \quad \text{for 70-volt lines, or:}$$

$$\text{Max } C = \frac{12}{Z_L} \quad \text{for any impedance output.}$$

At these limits, the reactance of the capacitance will be double the load impedance at 7 kc, and higher at lower frequencies. The limits may be doubled at a sacrifice of power capacity.

Conductor Size:

The size conductor to select for transmitting power over a distance depends upon the acceptable loss. The tabulated date Loss in Speaker Lines represents the length of the line for 1/2 db loss when transmitting various amounts of power at 70 volts. For each length, a conductor three size numbers smaller will have twice the resistance and will produce twice the loss (and will usually be bought for 70% the cost). For example, 40 watts can be carried 1200 feet on #16 wires with a loss of 1/2 db (12%), while #19 wire would cause a loss of 1 db (25%) under the same conditions. (Half of these 'losses' are not truly losses and can be recovered in some cases, as explained in the appendix). If a 40-watt amplifier provides the power at a cost of $120 dollars, the #19 pair wastes $30 worth of amplifier, while the #16 pair wastes just half this amount. If the costs of the wire are $28 and $20, respectively, the #16 would, in general, be the better choice. However, in any specific case, the difference of loss might require going to a larger amplifier, or there may be more amplifier power capacity than is needed, or it may seem wise to provide for future expansion, etc. Thus, various factors must be considered when choosing the wire size.

For transmitting power at line level (ranging from -20 dbm to +18 dbm, corresponding to 1/100 of a milliwatt to nearly 1/10 of a watt), the wire size is not important under a mile or two and, thereafter, we will be dealing with telephone lines. Technical Letter #103 gives the losses of a few common types of telephone lines.

A line settles down to long-line properties after its length reaches about 1 to 2 miles, depending upon the resistance and capacitance per unit of length.

VIII: CABLE CAPACITY

Cable capacity attenuates the high-frequencies and thus limits the permissible length of a cable run. The effect of a given capacity depends on the circuit impedance -- specifically, the parallel combination of the actual impedances looking both ways from the cable.

A 27-foot cable, having a capacity of 30 mmf per foot, will cause a 3 db loss at 10,000 cycles when connecting a 20,000Ω microphone to a pre-amplifier with the input circuit terminated in 100,000Ω. If the 10,000 cycle loss is to be limited to 1 db, the cable may not exceed a length of 13 feet. Table 3 gives a number of examples of cable lengths for a given loss and provides a formula for computing allowable cable length.

It is interesting to note that the capacity of 2-conductor shielded cable,in a circuit balanced to ground, is only about 2/3 of the value between one conductor and the other tied to shield. Consequently, balanced lines can be run 50% farther for the same high-frequency loss.

Capacity in Sound Systems:

Impedance (reactance)of a capacitor: X = 150Ω for 1 mfd at 1,000 cps (A formula to memorize)

Divide by frequency in kc for other frequencies.

Divide by capacity for other capacitors.

Examples: 2 mfd at 1,000 cps $= \frac{150}{2} = 75\Omega$

1 mfd at 10,000 cps $= \frac{150}{10} = 15\Omega$

1 mfd at 100 cps $= \frac{150}{.1} = 1{,}500\Omega$

1 mmf at 1,000 cps $= \frac{150}{1/1{,}000{,}000} = 150 \times 1{,}000{,}000 = 150$ megohms

(Note: The above are approximations; 6% low)

Effect of Capacity in a Circuit:

The effect of a given capacity depends upon it's reactance, whether it is in series or in shunt with the circuit, and the impedance of the circuit.

For capacity in series, the circuit impedance of significance (Rs) is the actual (not nominal or rated) impedance of the source, plus the actual impedance of the load. For capacity in shunt, the significant circuit impedance is the parallel impedance (Rp) of the actual source impedance in parallel with the actual load impedance. To determine the effect of the capacity, compute it's reactance and divide by Rs or Rp and read the loss from the following table:

X/Rs	X/Rp	Loss, db
0.1	10	Negligible
0.32	3.2	0.5
0.5	2.0	1.0
1.0	1.0	3.0
2.0	0.5	7.0
3.0	0.3	10.0

The table may also be used to determine the effect of an inductance, using X to represent the reactance of the inductance.

Length of Cable for 3db Loss at 10,000 Cps

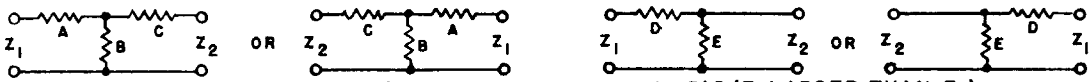
	30 mmf/ft	60 mmf/ft
1: Source: 150Ω Load: Very High — Rp = 150Ω Example: Low-Z Microphone to unterminated input transformer	3,000ft	1,500ft
2: Source: 600Ω Load: 600Ω — Rp = 300Ω Example: 600Ω attenuator to 436C with 600Ω resistor across input.	1,600ft	800ft
3: Source: 600Ω Load: Very High — Rp = 600Ω Example: As above, with 600Ω resistor omitted from 436C input	800ft	400ft
4: Source: 20,000Ω Load: 20,000Ω — Rp = 10,000Ω	50ft	25ft
5: Source: 20,000Ω Load: 100,000Ω — Rp = 18,000Ω Example: Hi-Z Microphone to amplifier input (no input transformer)	27ft	13ft

Formula: $L = \frac{15{,}000{,}000}{Rp \times C}$ Where C is capacity of cable in mmf per foot and Rp is parallel combination of actual impedances looking both ways from the cable,

Note 1: For 1 db loss at 10kc and 3 db loss at 20kc, reduce above cable lengths to half.

Note 2: For 2-conductor cables with one side grounded, use cable mfr's rated capacity for 'one conductor to other conductor tied to shield;' for 2-conductor cables in balanced circuit, use 2/3 this capacity.

DESIGN OF STANDARD PADS



T— PAD (Z_1 LARGER THAN Z_2)   L—PAD (Z_1 LARGER THAN Z_2)

Loss	VALUES OF PAD ARMS - OHMS 1000:1000 A	B	C	1000:500 A	B	C	1000:250 A	B	C	1000:100 A	B	C	Constants for Formula M	N	Max. Ratio Z_1/Z_2
1db	50	8640	50										8.69	8.64	1.01
2	100	4340	100										4.44	4.34	1.05
3	170	2830	170				IMPOSSIBLE						3.00	2.83	1.12
4	220	2110	220										2.33	2.11	1.23
5	280	1650	280					COMBINATIONS					1.93	1.65	1.37
6	340	1330	340										1.67	1.33	1.56
7	380	1220	380										1.50	1.12	1.8
8	430	950	430	710	670	20							1.38	.95	2.1
9	480	815	480	710	580	80							1.29	.815	2.5
10	520	700	520	720	500	110							1.22	.702	3.0
12	600	540	600	750	380	190	860	270	10				1.14	.54	4.4
14	660	416	660	790	290	250	870	210	60				1.08	.416	6.8
16	720	327	720	820	330	290	890	160	100	950	103	0	1.05	.327	10.4
18	780	256	780	850	180	330	900	130	120	950	80	20	1.03	.256	16.7
20	820	202	820	880	140	370	920	100	150	950	64	36	1.02	.202	25.4
25	900	113	900	920	80	420	950	60	190	960	36	64	1.01	.113	79.8
30	940	63	940	955	45	455	970	30	220	980	20	80	1.00	.063	247.
40	980	20	980	986	14	486	990	10	240	990	6	94	1.00	.02	2400.

NOTES:

1. In all cases Z_1 is larger than Z_2. For 500:1000, input is Z_2= 500, output is Z_1=1000 and column headed 1000:500 applies.

2. For Z_1 other than 1000 ohms, multiply A, B and C by $Z_1/1000$. (Examples: For 5000 ohms, multiply by 5., for 600 ohms multiply by .6)

3. To compute any possible pad use following formulae in which M and N are read from the table opposite the required loss:

   $A = Z_1\left(M - N\sqrt{Z_2/Z_1}\right)$ $B = Z_1 N\sqrt{Z_2/Z_1}$ $C = Z_1\left(M\, Z_2/Z_1 - N\sqrt{Z_2/Z_1}\right)$

   For loss values not given in the table, M and N may be computed as follows:

   $M = (K^2+1) \div (K^2-1)$ $N = 2K \div (K^2-1)$

   In these formulae K is the <u>voltage</u> ratio (larger than unity) corresponding to the loss in db. K may be read from the chart DB vs. Voltage Ratio.

4. Certain combinations of loss and impedance ratio cannot be obtained. The minimum value of loss is given by the 1st column when the impedance ratio is per column Z_1/Z_2.

<u>L PADS</u> are used to match two impedances with minimum loss. Leg D is in series with the <u>larger</u> circuit impedance, Z_1. E shunts the <u>smaller</u> circuit impedance, Z_2.

$D = Z_1\sqrt{1-Z_2/Z_1}$ $E = Z_2 \div \sqrt{1-Z_2/Z_1}$

The loss is read from the first column corresponding to Z_2/Z_1 in the last column.

Example: Match a 150 ohm source to a 600 load. Z = 600, Z_2 = 150. $\sqrt{1-Z_2/Z_1}$ = .87.
D = 600x.87 = 520 ohms; E= 150 ÷ .87 = 173 ohms. E shunts the 150 ohm circuit.

VOLTAGES FOR VARIOUS POWER LEVELS

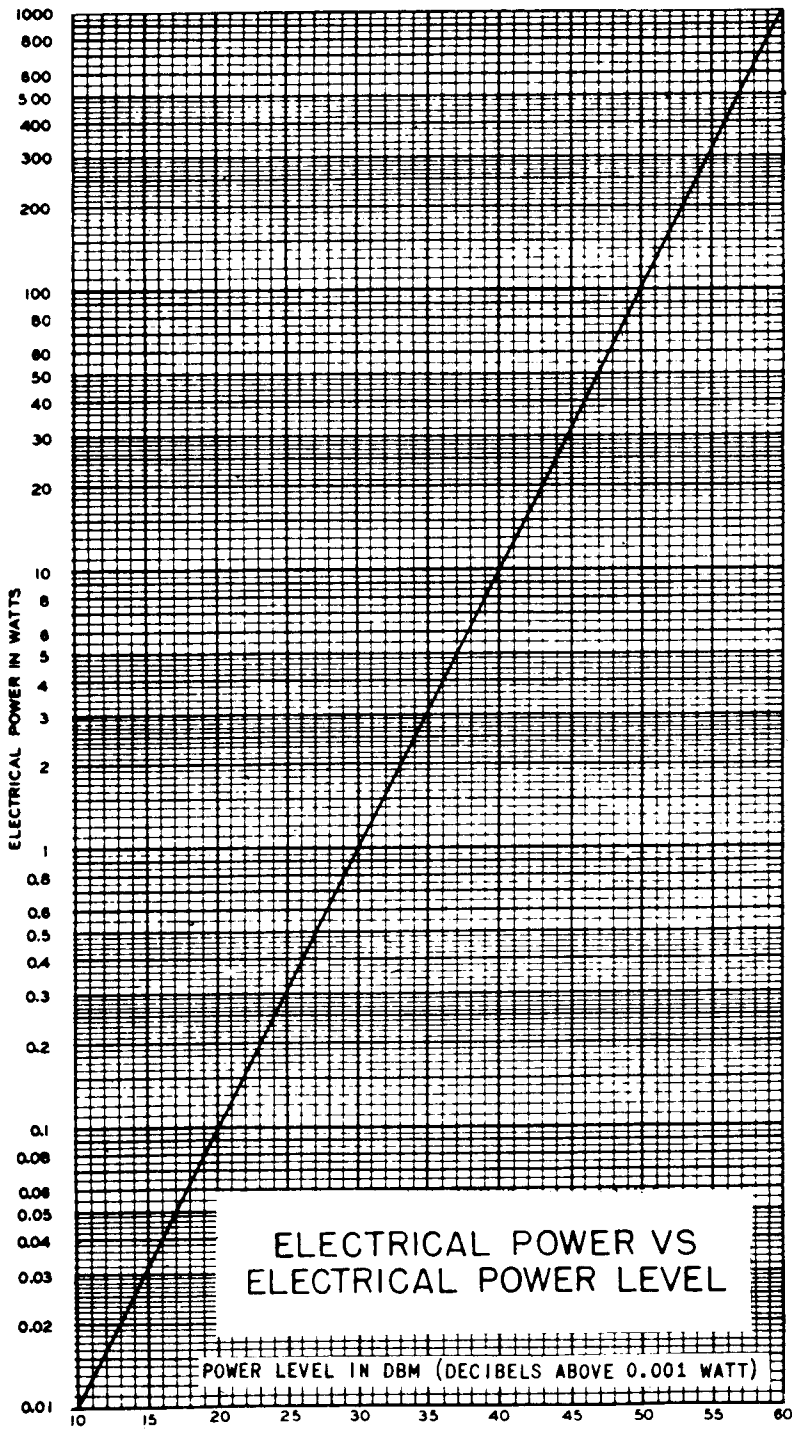
NOTE: For dbm of higher values, subtract 20, 40, or 60 db and refer to table below for result, then select appropriate decimal. Examples:

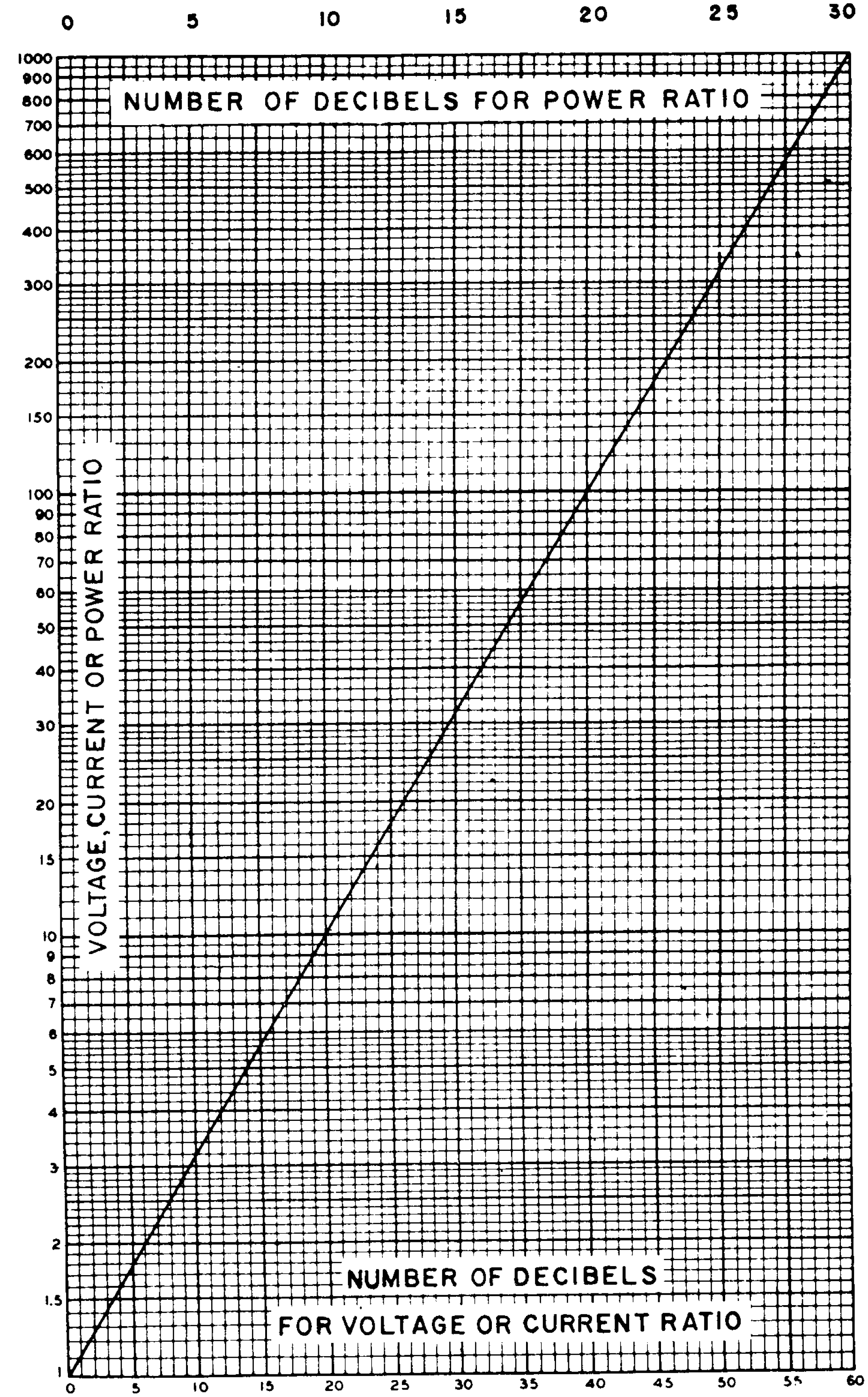
+22 dbm: (subtract 20) +2 dbm = 1.26v (1000 ohms),+22 = 12.6v (1000 ohms)
+32 dbm: (subtract 20)+12 dbm = 3.98v (1000 ohms),+32 = 39.8v (1000 ohms)
+42 dbm: (subtract 40) +2 dbm = 1.26v (1000 ohms),+42 = 126v (1000 ohms)
+52 dbm: (subtract 40)+12 dbm = 3.96v (1000 ohms),+52 = 398v (1000 ohms)

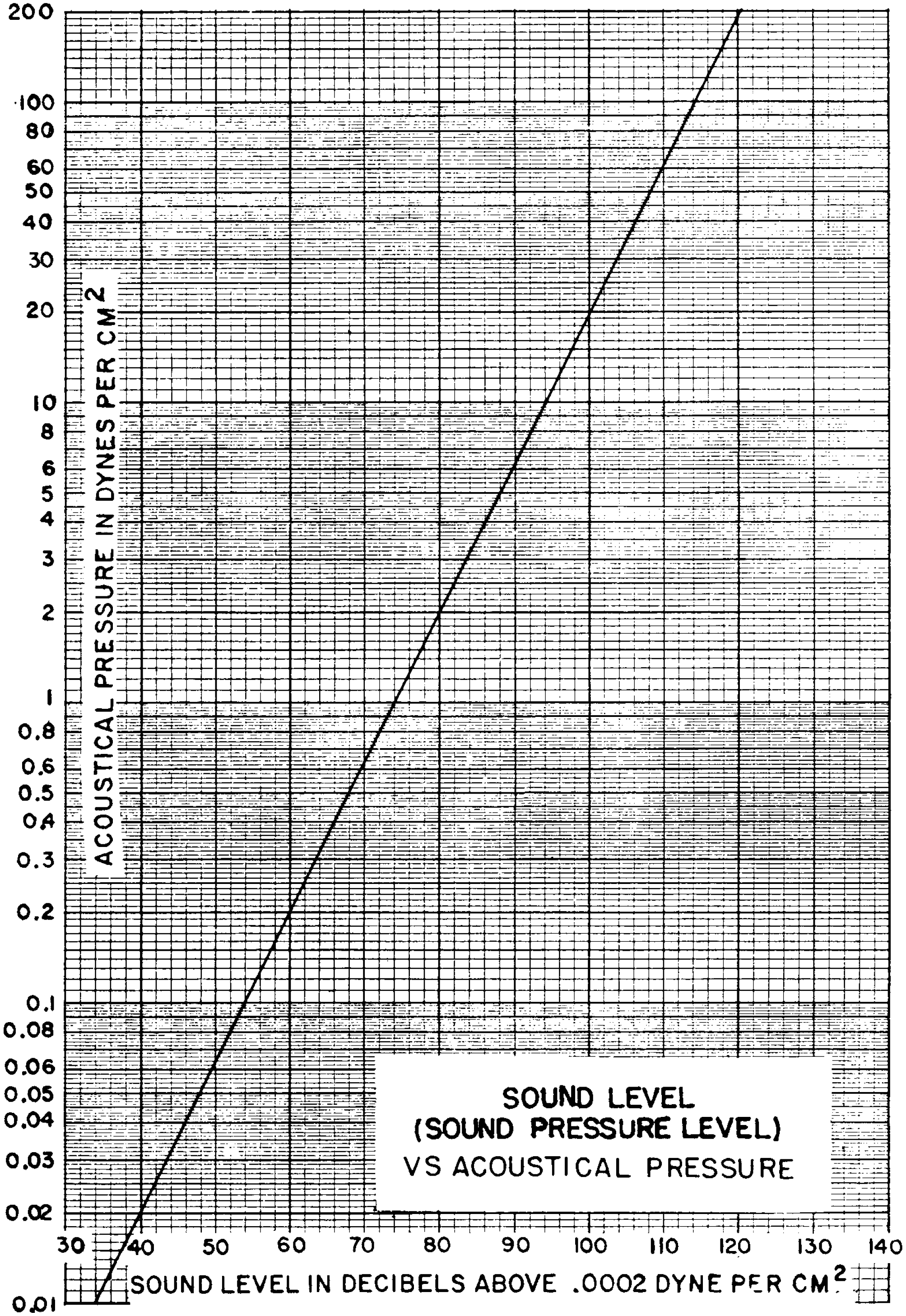
For dbm of lower values, reduce number by 20, 40, or 60 and refer to table below; then select proper decimal. Examples:

-22 dbm (reduce by 20) -2 dbm =.79v (1000 ohms), -22 = .079v (1000 ohms)
-32 dbm (reduce by 20)-12 dbm =.25v (1000 ohms), -32 = .025v (1000 ohms)
-42 dbm (reduce by 40) -2 dbm =.79v (1000 ohms), -42 = .0079v (1000 ohms)
-52 dbm (reduce by 40)-12 dbm =.25v (1000 ohms), -52 = .0025v (1000 ohms)

dbm	VOLTAGE (at listed impedance)									
	4	8	16	150	500	600	1000	5000	15,000	50,000
-20	.006	.009	.013	.039	.071	.078	.100	.224	.388	..707
-18	.008	.011	.016	.048	.089	.098	.126	.282	.488	.891
-16	.010	.014	.020	.061	.112	.122	.158	.354	.613	1.12
-14	.013	.018	.025	.077	.141	.154	.199	.446	.772	1.41
-12	.016	.022	.032	.097	.177	.195	.251	.562	.974	1.71
-10	.020	.028	.040	.122	.223	.245	.316	.708	1.22	2.23
- 8	.025	.035	.051	.154	.281	.308	.398	.892	1.54	2.81
- 6	.032	.045	.064	.193	.354	.388	.501	1.12	1.94	3.54
- 4	.040	.056	.080	.244	.446	.489	.631	1.41	2.45	4.46
- 2	.050	.070	.101	.307	.561	.615	.794	1.78	3.08	5.61
0.0	.063	.089	.127	.387	.707	.775	1.000	2.24	3.88	7.07
+ 2	.079	.112	.160	.480	.890	.977	1.26	2.82	4.88	8.91
+ 4	.099	.141	.201	.610	1.12	1.22	1.58	3.54	6.13	11.2
+ 6	.125	.177	.253	.770	1.41	1.54	1.99	4.46	7.72	14.1
+ 8	.158	.223	.319	.970	1.77	1.94	2.51	5.62	9.74	17.1
+10	.199	.281	.401	1.22	2.23	2.45	3.16	7.08	12.3	22.3
+12	.251	.354	.510	1.54	2.81	3.08	3.98	8.92	15.4	28.1
+14	.316	.446	.636	1.93	3.54	3.88	5.01	11.2	19.5	35.4
+16	.398	.562	.801	2.44	4.46	4.90	6.31	14.1	24.5	44.6
+18	.500	.707	1.01	3.07	5.61	6.15	7.94	17.8	30.8	56.1
+20	.630	.890	1.27	3.87	7.07	7.75	10.00	22.4	38.8	70.7
+22	.794	1.12	1.60	4.80	8.90	9.77	12.60	28.2	48.9	89.0
+24	.995	1.41	2.01	6.10	11.2	12.2	15.8	35.4	61.3	112
+26	1.25	1.77	2.50	7.70	14.1	15.4	19.9	44.6	77.2	141
+28	1.58	2.24	3.20	9.70	17.7	19.5	25.1	56.2	97.3	177
+30	1.99	2.81	4.01	12.2	22.3	24.5	31.6	70.8	122.6	223
+32	2.51	3.54	5.10	15.4	28.1	30.8	39.8	89.2	154.4	281
+34	3.16	4.46	6.36	19.3	35.4	38.8	50.1	112.2	194.3	354
+36	3.98	5.62	8.00	24.4	44.6	49.0	63.1	141.3	244.8	446
+38	5.00	7.07	10.10	30.7	56.1	61.5	79.4	177.9	308.0	561
+40	6.30	8.90	12.7	38.7	70.7	77.5	100.0	224.0	388	707
+42	7.94	11.2	16.0	48.0	80,9	97.7	126	282.2	488.8	890
+44	9.95	14.1	20.1	61.0	112	122	158	353.9	613	1117
+46	12.5	17.7	25.3	77.0	141	154	199	445.7	772.1	1406
+48	15.8	22.4	31.9	97.0	177	195	251	562.2	973.8	1775
+50	19.9	28.1	40.0	122	223	245	316	707.8	1226.1	2234


ELECTRICAL POWER VS
ELECTRICAL POWER LEVEL
ELECTRICAL POWER IN WATTS
POWER LEVEL IN DBM (DECIBELS ABOVE 0.001 WATT)
1000
800
600
500
400
300
200
100
80
60
50
40
30
20
10
8
6
5
4
3
2
1
0.8
0.6
0.5
0.4
0.3
0.2
0.1
0.08
0.06
0.05
0.04
0.03
0.02
0.01
10
15
20
25
30
35
40
45
50
55
60


NUMBER OF DECIBELS FOR POWER RATIO
VOLTAGE, CURRENT OR POWER RATIO
NUMBER OF DECIBELS
FOR VOLTAGE OR CURRENT RATIO
0
5
10
15
20
25
30
1000
900
800
700
600
500
400
300
200
150
100
90
80
70
60
50
40
30
20
15
10
9
8
7
6
5
4
3
2
1.5
1
0
5
10
15
20
25
30
35
40
45
50
55
60



LENGTH OF CABLE FOR 3 db LOSS AT 10kc

	30 mmf/ft	60 mmf/ft
1: Source: 150 ohms, Load: Very High } Rp=150Ω Example: Low-Z microphone to unterminated input XF*	3000 ft	1500 ft
2: Source: 600 ohms, Load: 600 ohms } Rp=300Ω Example: 600 ohm attenuator to 436B with 600 ohm resistor across input.	1600 ft	800 ft
3: Source: 600 ohms, Load: Very high } Rp=600Ω Example: As above with 600 ohm resistor omitted from 436B input.	800 ft	400 ft
4: Source: 20,000 ohms, Load: 20,000 ohms } Rp=10,000Ω	50 ft	25 ft
5: Source: 20,000 ohms, Load: 100,000 ohms } Rp=18,000Ω Example: Hi-Z microphone to amplifier input (No input XF*)	27 ft	13 ft

Formula: $L = \frac{15,000,000}{Rp \times C}$

*XF=transformer

where C is capacity of cable in mmf per foot and Rp is parallel combination of actual impedances looking both ways from the cable.

Note 1: For 1 db loss at 10kc and 3 db loss at 20kc, reduce above cable lengths to half.

Note 2: For 2/c cables with one side grounded, use cable mfr's rated capacity for "one conductor to other conductor tied to shield." For 2/c cables in balanced circuit, use 2/3 this capacity.

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