Engineering News

ALTEC LANSING

ALTEC LANSING 1515 S. Manchester Avenue, Anaheim, California

ALTEC DIVISION OF LING ALTEC INC

TWO PAPERS ON MICROPHONES

Technical Letter No. 148

MICROPHONES

William L. Hayes

From a paper presented at the Altec National Seminar, 1964.

INTRODUCTION

An essential part of any contemporary sound reinforcement system is a well-selected, high-quality microphone. Modern sound systems demand that the overall performance of the microphone be equivalent to that of the rest of the system. It is necessary that the microphone match the amplifier in frequency response, distortion and transient response; or that it be tailored to a specific response requirement. Purely acoustical characteristics such as the polar response in both the horizonal and the vertical planes must be satisfactory at all frequencies. If the microphone does not meet these high performance standards, then a substantial investment made in a quality amplifier and the other equipment is wasted money.

On the other hand, a well-chosen, high-quality microphone will pay off for the sound contractor.

But, in order to select the proper microphone, the sound contractor needs a practical knowledge of how they function.... how they are made....how to select them....and how to use them. He should know how to fit them into the proposed sound system....how to recognize their attributes and their limitations ....and -- if he makes the wrong choice -- he must know how to correct his mistake in a hurry.

BASIC TYPES

There are several basic categories of microphones. They are:

(1) omnidirectional (which includes the dynamic or moving coil pressure type and the condenser omnidirectional),

(2) bidirectional (which is the ribbon microphone, also known as the velocity type), and

(3) directional cardioid (unidirectional) which may be in the form of a dynamic (moving coil), ribbon, dual-unit type (dynamic and ribbon combined or, in some cases, two dynamics), and the condenser cardioid.

There are other types of microphones such as the carbon or the crystal, but they will not be discussed here except for comparative purposes.

OMNIDIRECTIONAL

Dynamic

The microphone which is dominant in the broadcasting and television industry and in public address systems is the dynamic type. An excellent microphone, it is often considered the "most microphone for the least money." If properly designed, it produces smooth frequency response, excellent sensitivity, and low distortion. A rugged, self-contained unit (i.e., it requires no power supply), it is inexpensive to manufacture.

In a dynamic pressure unit (Figure 1), the magnet and its associated parts (Magnetic return, pole piece, and pole plate) produce a concentrated magnetic flux of approximately 10,000 Gauss in a small gap. The magnetic circuit is designed for optimum flux density. This condition is verified by empirical flux density measurements.

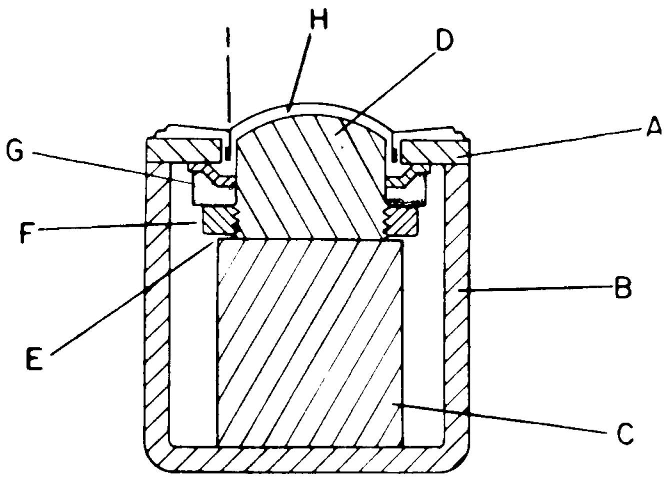
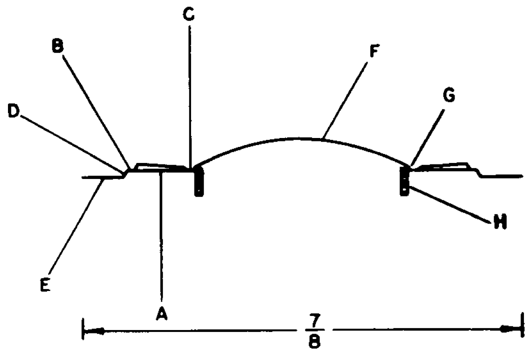


FIGURE 1 - Dynamic Pressure Unit

A- pole plate
B- magnetic return
C- Alnico V magnet
D- pole piece
E- threaded portion
F- threaded adjusting ring
G- acoustic resistor (felt)
H- capacitance under diaphragm dome
I- voice coil



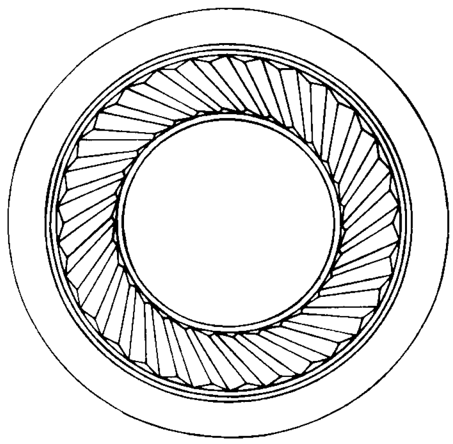


FIGURE 2 - Omnidirectional Diaphragm and Voice Coil Assembly

A- tangential compliance section	E- cementing flat
B- hinge point	F- dome
C- hinge point	G- coil seat
D- spacer	H- voice coil

The diaphragm, a key item in the performance of a microphone, supports the voice coil centrally in the magnetic gap, maintaining that center condition with a gap of only six mils (0.006 inches) on either side of the voice coil.

An Altec omnidirectional diaphragm and voice coil assembly is shown in Figure 2. The compliance section (A) has two hinge points at (B) and (C). Between these points is a section made up of tangential corrugations; triangular sections which stiffen this portion. The hinge points are designed to permit high compliance action. A spacer (D) supports the moving part of the diaphragm away from the top pole plate to provide room for its movement. (E) is a cementing flat which is bonded to the face plate. A dome (F) is another stiffness section with a radius designed to provide adequate acoustical capacitance. The coil seat (G) is a small step which permits the voice coil (H) to be mounted and centered on the diaphragm, and which provides an area for a bond to unite with the diaphragm.

The general configuration shown in Figure 2 is typical of most modern microphones and the diaphragm designs of some speaker drivers (tweeters). Although it is a conventional design, many hours were spent developing a diaphragm which would provide optimum results in smoothness of response, maximum bass response, and maximum high-frequency response.

Until recently, most microphone diaphragms were made of aluminum and usually were less than 1 mil (0.001 inch) in thickness. Aluminum, once considered the best material available, is light in weight, fairly easy to form and it maintains its dimensional stability after it is molded. Unaffected by extremes in temperature or humidity, aluminum, in general, serves the purpose. But that one mil thickness makes the diaphragms fragile; a fragility which has ruined many dispositions. When it is touched or otherwise deformed by excessive pressure, an aluminum diaphragm is dead.

A great deal of research was performed in an attempt to find another material which had the acoustical advantages of aluminum, but yet was tough enough to withstand abnormal deformation. The material turned out to be a polyester film manufactured by the Dupont Company and known by its tradename, Mylar®.

Mylar® is a unique plastic. Extremely tough, it has high tensile strength, high resistance to wear, and outstanding flex life. Capable of withstanding high temperatures, it is used in motors, transformers and in other critical applications. Mylar® diaphragms have been tested with temperature variations from -40 to +170 degrees Fahrenheit, cycled over long periods, without any impairment to the diaphragm. Since Mylar® is extremely stable, its properties will not change within the temperature and humidity range in which microphones are used.

The specific gravity of Mylar® is approximately 1.3 (as compared to 2.7 for aluminum), therefore, a Mylar® diaphragm may be made considerably thicker without upsetting the relationship of the diaphragm mass to the voice coil mass. (In the case of the omnidirectional diaphragm, the thickness is 1-1/2 mils.) Mylar® diaphragms are formed under high temperature and high pressure, a process in which the molecular structure is formed permanently to establish a 'dimensional memory' which is highly retentive. Unlike aluminum, Mylar® diaphragms will retain their shape and dimensional stability although they may be subjected to drastic momentary deformations. Such a diaphragm may be bent at right angles and, when allowed to spring back, will still operate. It may not remain acoustically perfect, but the microphone will continue to function. Diaphragms have been flattened against the supporting structure by pressure from the palm of a hand and yet -- when released -- the microphone functioned properly.

Altec uses a semi-automatic machine to form the Mylar® diaphragms. A female-type mold conforming to the proper configuration is heated to approximately 400 degree Fahrenheit. The Mylar® blank is pressed against the mold with a silicone rubber pad (silicone rubber will withstand the high temperatures required), then pressure is applied to the press. This pressure is about 320 pounds-per-square inch or, in terms of force, about half a ton. While pressure is maintained, the mold is cooled to room temperature, then the finished diaphragm is extracted from the mold and checked for quality of dimensions and form.

The voice coil, the controlling part of the mass involved in the diaphragm and voice coil assembly, weighs more than the diaphragm. To achieve a good high frequency response, the material used must be of minimum weight. Aluminum wire has the best mass-to-conductance ratio and provides the unit with a frequency response extending up to 20,000 cycles. Because

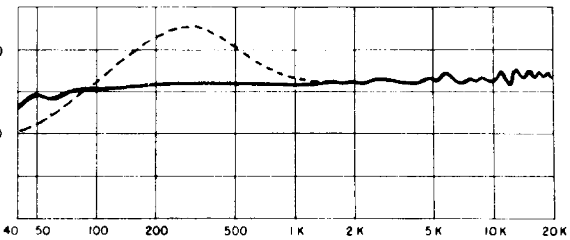


FIGURE 3 - Diaphragm and Voice Coil Assembly, Response Curve

the voice coil and diaphragm assembly has mass (analogous to inductance in an electrical circuit) and compliance (analogous to capacitance), the assembly will resonate at a given frequency in the manner of a tuned electrical circuit. The 'free-cone' resonance of an undamped unit is approximately 350 cycles per second.

If it were left in this undamped state, the response of the assembly would be as in indicated by the dotted line in Figure 3. The resonant characteristic is damped out by an acoustic resistor, a felt ring which covers the openings in the centering ring behind the diaphragm. This is analogous to an electrical resistance in a tuned circuit and damps the resonant point down to a flat response. The acoustic resistance is controlled by a threaded ring behind it which changes the pressure on the felt ring to alter the acoustic resistance.

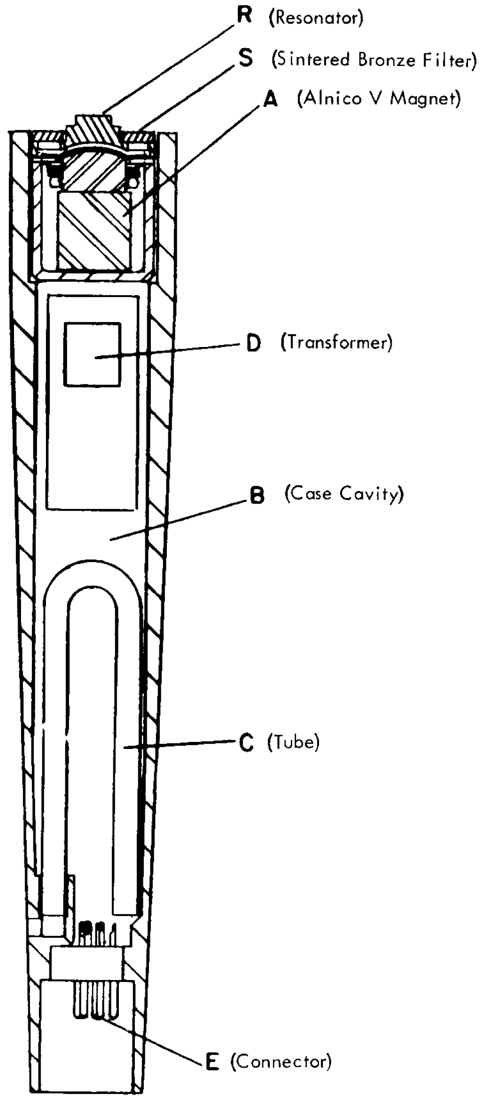


FIGURE 4 - Omnidirectional Microphone (cross-section)

Even with the unit damped, there is a drooping in the lower frequency range from about 200 cycles down (dotted curve, Figure 3). This is corrected by the use of additional acoustic resonant devices inside the microphone case. A cavity behind the unit (analogous to capacitance) helps resonate at the low frequency with the mass (inductance) of the diaphragm and voice coil assembly. Another tuned resonant circuit is added in the form of a tube (C, Figure 4) which couples the inside cavity of the microphone housing to the outside. This tube has an acoustic inductance which is tuned to a low frequency (in this case, 50 cycles) so that a flat response extending down to 35 cycles may be obtained.

The radius of the dome provides stiffness and an acoustic capacity in the form of an air cavity between the diaphragm and the dome of the pole piece. This capacitance resonates with the mass (inductance) of the assembly and helps extend the response up to the 20,000 cycle area.

To control this resonance, an acoustic resistance in the form of a sintered bronze filter is placed in front of the diaphragm (S, Figure 4). This filter also serves as an effective protection device.

Sintered bronze, a recent development in metallurgy and a great asset to acoustic devices, is composed of bronze spheres which are pressed to form a porous sheet of material. Used as a filter in most Altec microphones, it prevents dirt particles, magnetic chips and moisture from gravitating to the inside of the unit. Magnetic chips, if allowed to reach the magnetic gap area, eventually will accumulate on top of the diaphragm and impair the frequency response. It is possible for such chips to "pin" the diaphragm to the pole piece and thereby render the microphone inoperative.

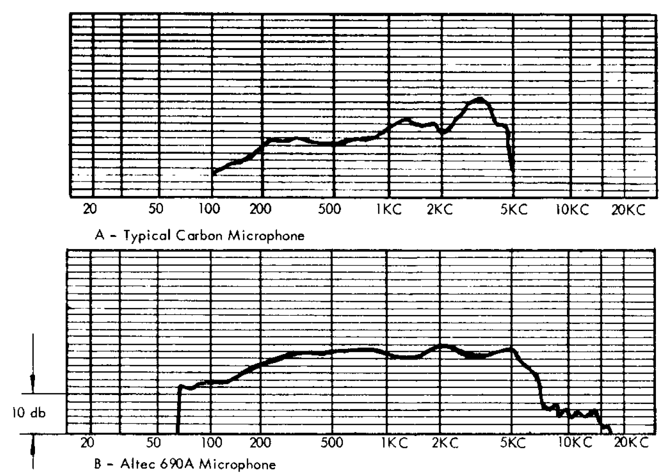


FIGURE 5 - Handset Microphone Response at 0° Incidence, 1/4-Inch From Artificial Mouth

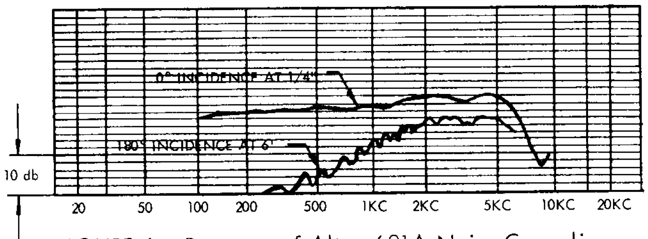


FIGURE 6 - Response of Altec 691A Noise Canceling Microphone in Boom Headset (with artificial mouth)

The Altec 690A, a miniaturized dynamic omnidirectional microphone and amplifier unit, is designed to fit standard telephone handsets. Because the sensitivity is boosted by a transistor amplifier, the microphone may be made smaller. The amplifier has a polarity reversal feature which allows dc voltage, simplexed over the line to the microphone, to be reversed without interfering with the operation of the microphone or the amplifier.

A typical carbon button microphone will have a peak at around 3000 cycles (Figure 5A), while the 690A has a rising characteristic which is much smoother (Figure 5B).

Also, Altec has noise canceling versions of boom-type operator headsets; the 691A, a 24-volt unit, and the 692A, a 48-volt unit. (The 693A and 694A are kit forms of these two microphones.) Amplifier specifications for these units are the same as for the 690A, and the response characteristics are shown in Figure 6.

Condenser

The condenser microphone usually is considered the ultimate in microphone design because of its smooth, precise frequency response, its low distortion and its excellent transient response. Operation of the condenser microphone depends upon the variation of capacity between two plates, one movable and the other fixed. With a direct current polarization voltage applied across these plates, the moving diaphragm alters the standing voltage and causes the required electrical signal output. The signal from the condenser capsule is fed directly into a cathode follower which, for capacitance reasons, is contained in the same case as the condenser capsule. Altec makes condenser microphones in both omnidirectional and cardioid types.

BIDIRECTIONAL MICROPHONES

Ribbon

A ribbon (velocity) microphone is bidirectional; that is, it picks up at the front and the back (180° axis) but it attenuates on either side. The ribbon, a thin strip of aluminum foil, is suspended between two poles of a permanent magnet system. When the ribbon is vibrated by sound pressure, it acts as a conductor; cutting the lines of force created by the magnet and the gap, and develops voltage. This thin ribbon makes the microphone so fragile that blasts of pressure may stretch it, altering its response or, in some cases, causing it to fail. The ribbon microphone primarily is a studio or inside unit. It should be suspended on a stand or boom in an area where it is not subjected to the wind or to extreme shocks.

DIRECTIONAL CARDIOID MICROPHONES

The (uni)directional cardioid microphone is a rugged unit used in broadcasting and in public address systems. The dynamic

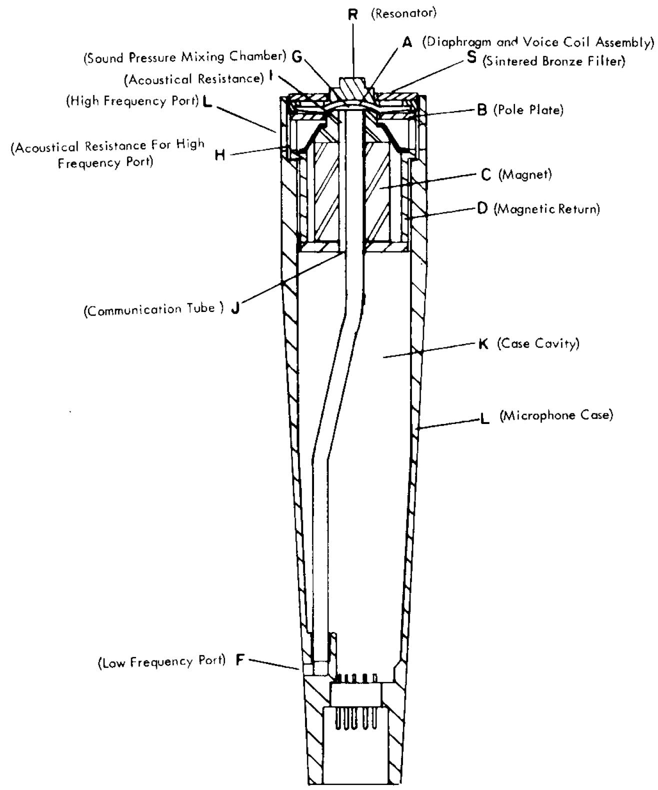


FIGURE 7 - Representative Cardioid Microphone (cross-section)

pressure unit of the cardioid is more complicated than that of the omnidirectional. Although a flat extended frequency response is required, it is equally important that the back response ('discrimination') be as great and as uniform as possible throughout the frequency spectrum. Therefore, a cardioid must have rear sound entrances and acoustical phase-shifting networks as well as many of the operational features found in an omnidirectional microphone.

Figure 7 represents a cross-section of an Altec cardioid microphone in which some components (such as the transformer) have been eliminated in order to simplify the illustration. The diaphragm and voice coil assembly (A) is much the same as for the omnidirectional. The low frequency port (F) terminates in a sound pressure mixing chamber (G) beneath the diaphragm. A communication tube (J) leads from the back of the diaphragm area to the main cavity (K).

Basically, the cardioid microphone operates on the principle that if the pressure at the front of the diaphragm is the same as the pressure at the rear, cancellation will occur. Figure 8 is a simplified diagram and an electrical analog representing the mechano-acoustical circuit as it operates in a cardioid microphone. When a cardioid microphone is pointed toward a sound source, sound pressure is presented to the front of the diaphragm (P1, Figure 8). Simultaneously, pressure travels around to enter the side ports (P2) and the rear ports (P3). However, P2 is out of phase with P1 so a pressure gradient (difference) is created which actuates the diaphragm causing a voltage output. After entering the rear port, the third pressure (P3) travels through the tube where it is delayed by an acoustic impedance before it creates another pressure gradient. These acoustic impedances

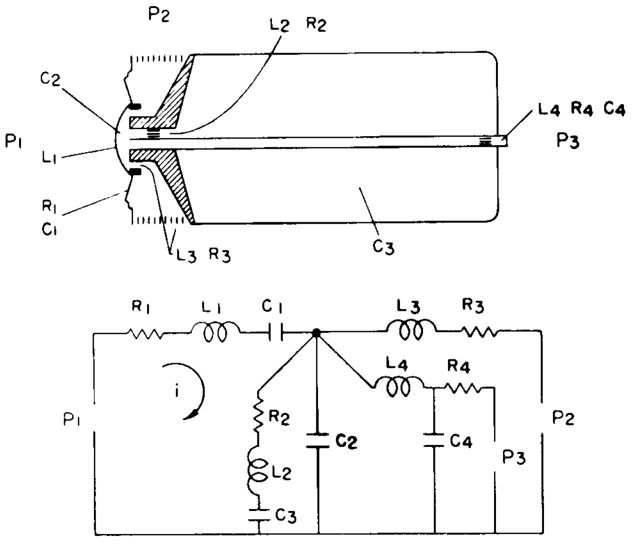


FIGURE 8 - Simplified Diagram and Electrical Analog of the Mechano-Acoustical Circuit of Cardioid Microphone Operation

are designed to provide smooth flat frequency response throughout the audible spectrum so that a tailored frequency response is attained.

With the sound source at the rear of the microphone, the low frequency sound enters the low frequency tube and creates a pressure (P3). At the same time the sound travels around the outside of the microphone and strikes the diaphragm from the front (P1). For the low frequency range up to about 600 cycles the acoustic impedance of the tube delays the sound so that it strikes the diaphragm at the same time and, essentially, at the same phase as the sound arriving at the front. Because the force on the front of the diaphragm is equal to the force at the rear, cancellation occurs and a null is created.

From 600 or 700 cycles on up to the high frequency end, the sound pressure enters the side ports (P2). This sound also goes around the outside of the case and enters the front (P1). The acoustic impedances of the side ports are such that the sound is presented in essentially the same phase relationship as the sound arriving at the front of the diaphragm. As a result, the two pressures tend to cancel out each other and discrimination occurs.

ALTEC MICROPHONES

Before Altec developed a new series of microphones, the company sponsored a survey to determine what features would be desirable. The results of this canvas indicated that a majority of sound engineers preferred a slight tilt in the response in order to accentuate the higher frequencies, and they agreed that the average of this tilt should be about four decibels in the range from 100 to 10,000 cycles.

Omnidirectional

Altec microphones 681A, 682A, 684A (Figure 9) and 686A are designed with such a tilt, providing a high degree of articulation in these models which are used primarily for voice purposes.

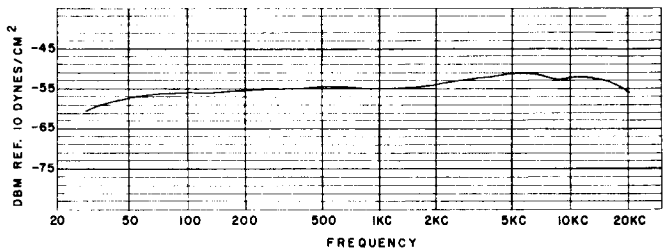


FIGURE 9 - Response Curve, Altec 684A Microphone

A subsequent survey disclosed requests for an omnidirectional microphone with an extremely flat response (no tilt). Altec complied by designing the 688A, an omnidirectional microphone which has a flat response extending up to 20,000 cycles (see Figure 10). This flat response is useful in studio work where music is being recorded or broadcast, whereas the tilted response is applicable to public address systems or broadcast work where voice reproduction is predominant.

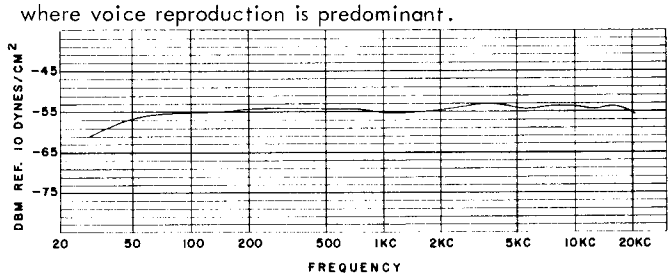


FIGURE 10 - Response Curve, Altec 688A Microphone

The Altec 687B, another dynamic omnidirectional microphone, is called the "announce" microphone since it has many features useful in a paging type of system. It is designed to be either hand-held or mounted on a stand, has a push-to-talk switch, both low and high impedance, and a rugged die-cast, chrome-plated case. Components include the omnidirectional pressure unit incorporating the Mylar® diaphragm and a sintered bronze filter.

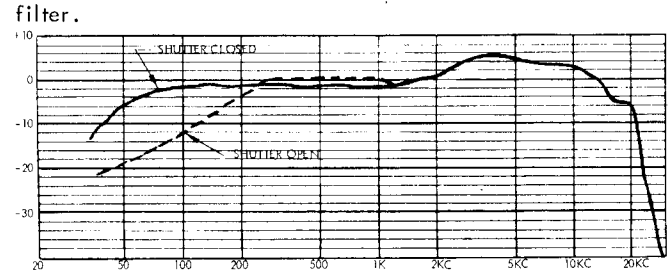


FIGURE 11 - Response Curve, Altec 687B Microphone

The frequency response of the Altec 687B (Figure 11) is tilted with a peak in the 4000 to 5000 cycle area as is generally preferred for a paging type of microphone. Also, the 678B has a variable bass frequency response control which permits alteration of the tilt in the lower audio range, making the microphone adaptable for the best articulation in any acoustical situation.

The Altec 686A, a lavalier-type microphone, is small and lightweight, permitting freedom of action for the wearer. The case is made of die cast aluminum instead of the usual zinc. In a lavalier-type microphone, which is suspended from the neck and rests against the chest of the speaker, there is an accentuation of bass response due to the conduction of sound from the chest cavity. To compensate for this bass response, the lavalier has a bass response roll-off starting at about 500 cycles. (See Figure 12.) Also, the general response from 500 cycles up is tilted upwards with an average slope of four decibels from 100 up to 10,000 cycles. Since the 686A is strictly a voice microphone, this tilt is a desirable feature.

The general construction of the lavalier, shown in Figure 13, is similar to the regular omnidirectional microphone except for the size of the case. To provide the bass roll-off, the design incorporates a smaller cavity inside the case and omits the resonator tube.

The Altec M-20 system, which uses the 21D condenser microphone, has an extended and smooth response down to 10 cycles, and up to 15,000 cycles. Small in size, the uniformity of its omnidirectional pickup makes it ideal for recording and broadcast requirements while its ruggedness and freedom from magnetic pickup are features which are helpful in industrial and outdoor applications. It also may be used as a lavalier.

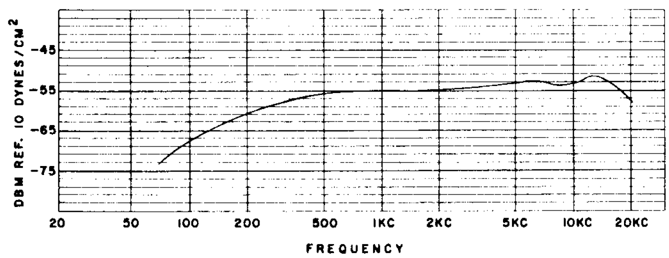


FIGURE 12 - Response Curve, Altec 686A Microphone

Bidirectional

The Altec 639A and 639B models are multi-pattern microphones which have a long record of acceptance from the audio industry. Using dual units -- dynamic and ribbon -- these microphones have a uniform response from 40 to 10,000 cycles. Their outstanding feature is the versatility of their directional patterns. The 639A may be set for cardioid and omnidirectional pickup as well as for bidirectional. The 639A permits three additional settings for various degrees of rear sensitivity.

TESTIMONIALS FOR THE ALTEC M-20 AND M-30 MICROPHONE SYSTEMS

....At St. John's Methodist Church in direct comparison tests during an actual service, the much greater sensitivity, clarity and extremely sharp articulation of the Altec M-30 microphone was alternately demonstrated against an Executone system. On the Executone system feedback occurred before a truly adequate reinforcement level could be reached, but the Altec system was able to reach a level considerably in excess of the requirements before any feedback occurred. And the Altec system sold at a price some 40 percent higher.

....The Falk Theater and Henry Grady School Auditorium use Altec M-20 systems suspended from overhead. Pickup of the entire stage performance is accomplished with remarkable clarity and boost in level without any feedback.

....At the Citrus Commission Auditorium in Lakeland, Florida, a single Altec M-20 is being used to record the meetings with seventeen commissioners on stage, plus all the audience participation over the entire auditorium. Previously seventeen microphones were used on the stage and there were no arrangements for the audience pickup. The signal-to-noise ratio is surprisingly good for the amount of gain used. Voice pickup is clear and intelligible, and the proceedings have been broadcast. No other microphones used would pick up the audience since the noise factors were too great to be usable. A single Altec M-20 now is used to record all sessions.

....At MacDill Field, there are two installations in Strike Command Headquarters; one in the War Room and one in the briefing room. The original War Room used one Altec M-30 for use of the general only. The other microphones were all Altec 661C's. The General ordered all microphones changed to the M-30 because of the remarkable pickup characteristics and speech quality of the microphone. Much higher levels were obtained before feedback occurred in spite of the fact that the microphones were suspended from the ceiling and were directly in the path of the Altec 755C loudspeakers which are installed in the ceiling.

The new briefing room also uses 755C speakers which are mounted in sloping baffles on either end of the room. Twenty Altec M-20 microphones are used and all are activated at the same time that sound is being reinforced in the room.

....Every church where an Altec M-20 or M-30 microphone system has been demonstrated in comparison with any competitive installation has wound up ordering the Altec regardless of the price differential. Wherever dissatisfaction with pickup at an alter or a pulpit has occurred, use of a M-20 or a M-30 has eliminated the complaint.

....First Baptist Church in Tampa had seven EV and Shure microphones and did not want additional microphones when their system was revamped. But when the Altec M-30 was demonstrated in direct comparison with others, the committee chairman said, "If it cost $1000, we would have to have it."

....A theatre in Montgomery, Alabama had five EV cardioids in the footlights. The system was unworkable; had very little gain. They tried one Altec M-30 at 7 to 10 decibels gain working off the microphone. Then they suspended two M-30's seven feet apart some seven or eight feet high, tilting them 45 degrees down. The microphones could not be seen. The report was that "previously nothing could be heard -- now one can hear anything on the stage."

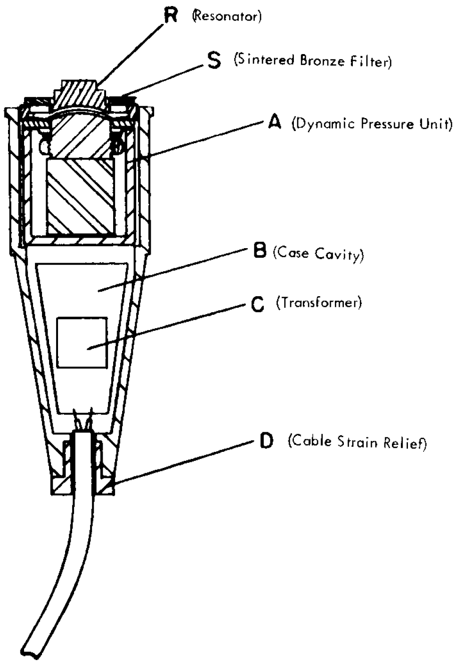


FIGURE 13 - Cross-section, Altec 686A Microphone

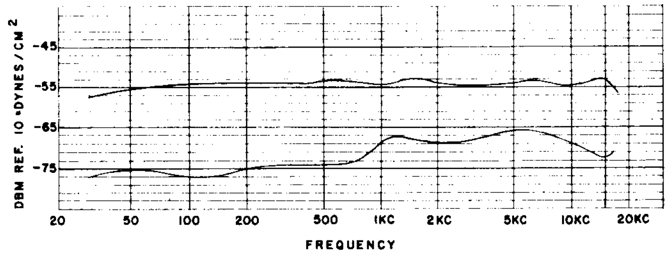


FIGURE 14 - Response Curve, Altec 689A Cardiod Microphone

Both the Altec 685A and 683A cardioid microphones, designed for use in voice applications, incorporate a four decibel tilt in the front response. An effective magnetic shield surrounds the transformer network and a hum-bucking coil is mounted inside the pressure unit in the 685A and 689A microphones. This combination reduces hum pickup in the studio or other areas where high 60-cycle flux densities may exist.

The Altec M-30 system uses the 29B condenser cardioid microphone. This microphone has a smooth response (Figure 15) from 20 to 20,000 cycles, and a wide frontal pickup pattern which extends 45 degrees to either side. The microphone has a high discrimination at 180 degrees, and especially deep discrimination in the 400 to 4000 cycle area. The 29B, a miniature directional microphone, is capable of translating the entire frequency and dynamic range without false accentuations.

The Altec 695A NCD (Noise Canceling Dynamic) microphone is designed for communications work -- primarily in telephone installations, mobile applications and aircraft radio usage. It is equally adaptable for paging systems wherein a high ambient noise situation exists. A close-talking, dynamic type in combination with a transistor amplifier, the 695A is designed to replace the majority of carbon button units and thereby increase the articulation and intelligibility. Power for the amplifier is taken directly from the supply which formerly powered the carbon unit. As a result, it serves as a direct replacement for such units.

Cardioid

The frequency response characteristics and basic curve for the Altec 689A cardioid microphone, known as the music cardioid, are shown in Figure 14. The curves show the flat response and the back response of the microphone to indicate the discrimination obtained throughout the 30 to 16,000 cycle range. Each 689A is supplied with a calibration curve made with Bruel and Kjaer test equipment in the Altec anechoic chamber. Tests made in this chamber represent the most accurate recording of microphone responses obtainable. Altec is one of the few companies which provides this type of service with its professional microphones.

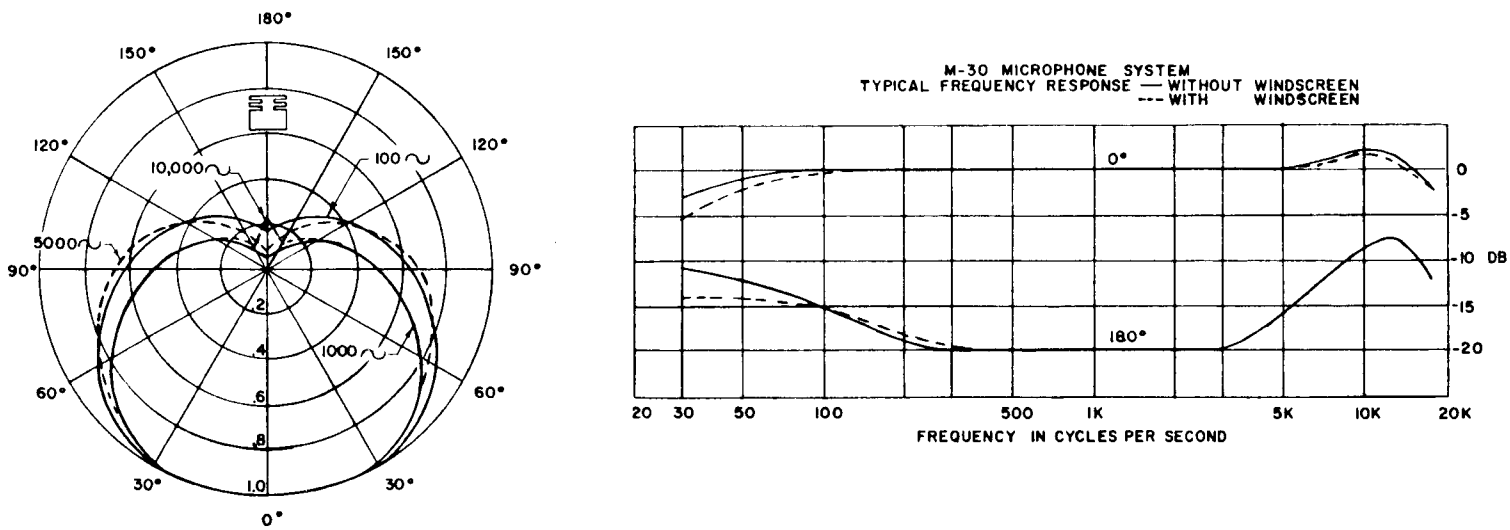


FIGURE 15 - Polar And Frequency Characteristics of the Altec M-30 Microphone System

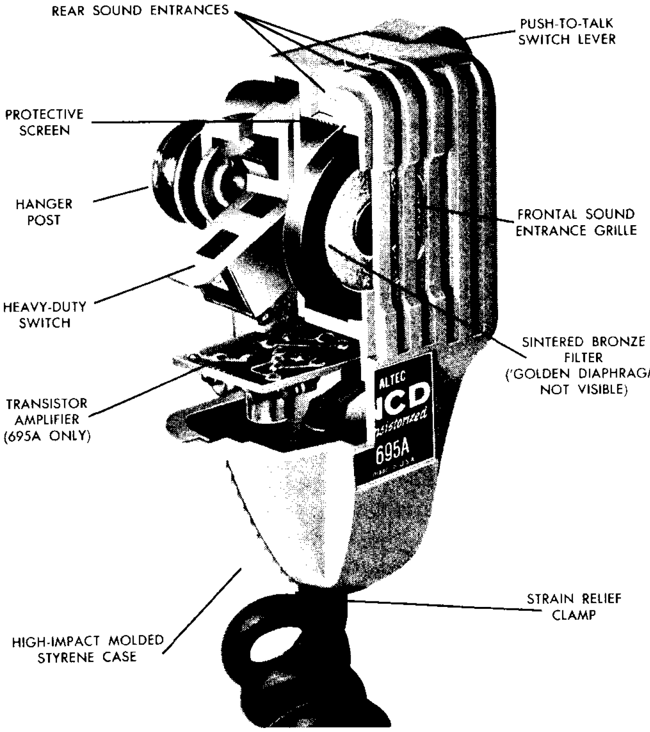


FIGURE 16 - Cutaway Photograph of the Altec 695A Microphone Showing Component Parts. (Note: The Altec 696B Microphone is identical with the 695A except for omission of the transistor amplifier.)

A cross-section of the Altec 695A is shown in Figure 16. The sintered bronze filter is directly behind the grill and in front of the Mylar® diaphragm of the dynamic unit. The unit is a miniaturized version of the omnidirectional type except that it is noise-canceling and enclosed inside the space behind the sintered bronze filter. The rear sound ports on top of the unit are loaded with acoustic resistance. The front resistance (due to the filter) and the rear port resistances are selected to cancel sounds coming from a distance, from the rear and from the front of the microphone which is designed to be used approximately 1/4-inch from the lips. The sound source creates a spherical wave shape which impinges on the front of the diaphragm through the sintered bronze filter. The sound also travels the longer distance over the top to the sound ports. This distance is about three times greater than the distance from the lips (point source) to the front entrance. This difference creates a pressure gradient which causes the diaphragm to actuate in accordance with the speech waves. Since the majority of all noise is of a random nature, a wave front approaching the rear of the microphone is a plane wave; that is, relatively flat across its front. Therefore, there is almost no difference in pressure between the wave front at the rear port and the wave at the front port. This is because the distance between the front and rear ports is so short as compared to the long distance between the noise source and the microphone. Since the pressure on the front and on the back of the diaphragm are virtually the same, the diaphragm is not actuated and the noise is canceled.

Basically, the same thing occurs when the 695A is pointed toward a sound source. However, the discrimination against the noise is not quite so great because the microphone has a cardioid characteristic in its response.

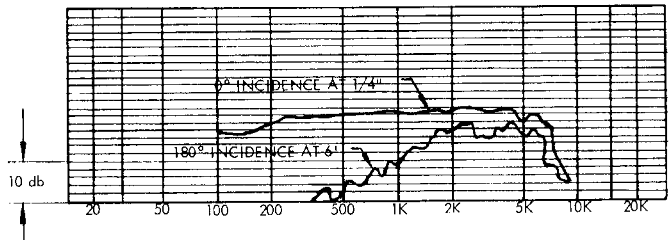


FIGURE 17 - Response Curve, Altec 695A and 696B Microphones

The Altec 696B is a transformer version of the 695A and has a standard microphone transformer replacing the transistor amplifier. This transformer changes the voice coil impedance of the unit up to 200 ohms, or to a high impedance of 20,000 ohms. The response curve for both the 695A and the 696B (Figure 17) shows the zero incidence response at a quarter-inch from the sound source, and the 180-degree incidence response at six feet. It is indicative of the excellent noise-canceling characteristics of these microphones.

Instrumentation Microphones

Altec's instrumentation condenser microphones, the 21B series, are intended for high intensity sound measurement. These are rugged instruments designed for practical measurements of such high sound pressures as the output of jet engines and missiles. The microphones have high accuracy, low distortion, stable construction, resistance to temperature and altitude effects, and they maintain accurate calibration. Models are available to measure SPL (Sound Pressure Level) from 150 to 200 decibels re 0.0002 dyne/CM^2. This range is covered by the basic group consisting of:

21D	-	150 db linear limit
21BR150	-	158 db linear limit
21BR180	-	170 db linear limit
21BR200	-	185 db linear limit
21BR220	-	200 db linear limit

Seventeen combinations of each of these models (making a total of 68 variations) are available for a variety of requirements on special order.

21BR-1	Basic microphone
21BR-2	Undamped, fast vent
21BR-3	Probe microphone
21BR-4	Electro static actuator
21BR-5	Open face
21BR-6	Pin hole, flush mount, undamped, fast vent
21BR-7	Flush mount, bronze filter face, undamped, fast vent
21BR-8	1/8" dia. probe, open face, probe 3-5/16" long, fast vent
21BR-9	-3 with 45° bend in probe (NASA, Cleveland)
21BR-10	-3 with 1/8" o.d. probe, fast vent (NASA, Langley Field)
21BR-11	-10 with 45° bend in probe (NASA, Langley Field)
21BR-12	Open face, fast vent
21BR-13	Dummy
21BR-14	Undamped, bronze filter (General Radio)
21BR-15	Undamped, pin hole off center
21BR-16	-3 with fast vent
21BR-17	Pin hole (0.266" dia.), sintered bronze filter

NOTE: FV suffix denotes front vent (not applicable to -2, -6, -7, -8, -10, -11, -12, or -16)

WIND AND "POP" SCREENS

Cardioid microphones, designed to have a flat response down to 30 cycles, are susceptible to the "pops" and "poofs" which are caused by speakers talking too close to the microphone. A large breath blast is created when a word containing the "P" or "B" sound is spoken. Essentially, this is a low frequency blast which causes the microphone diaphragm to move wildly. In turn, this creates a large "pop" sound in the entire system.

One method of eliminating this noise involves a rolling-off of the lower frequencies. However, since Altec cardioids are designed to be as flat as possible down to 30 cycles, an acoustical solution was sought; one which would not alter the over-all response of the microphone. The problem was solved with the "pop" screen now featured in Altec's cardioid microphones.

The Altec "pop" screen is a simple device made of two wire-mesh screens treated with flocking material to create an acoustical resistance. This so-called two-stage filter, a vast improvement to the overall microphone system, has replaced the rosette fronts on the cardioid microphones. (To maintain a similar appearance, rosettes have been replaced with wire-mesh nosepieces on all Altec microphones except the 686A lavalier.)

For some time the wind noise in sound systems used for outdoor events has created problems for sound engineers. Wind screens were designed, but they were not interchangeable between various microphones. In most cases, improvised screens were unsatisfactory and tended to attenuate the high frequency response or, in the case of the directional microphones, upset the polar response.

A simple but effective polyester foam screen is available to fit the 681A, 682A, 684A and 688A dynamic omnidirectional microphones, and a larger model is designed for the 683A, 685A and 689A cardioid microphones. Providing approximately 11 decibels of wind noise attenuation, these Altec screens (which are free) will not upset the discrimination of the cardioid pattern.

Also, there is the Altec 170B wind screen for the M-20 and M-30 systems. It has approximately 24 decibels of wind noise reduction without deterioration to the high frequency response or, in the case of the M-30, the cardioid discrimination (Figure 18).

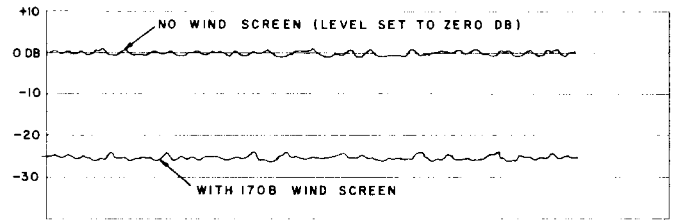


FIGURE 18 - Response Curve, Altec 170B Wind Screen Mounted on the Altec M-30 Microphone System

The 170B wind screen also serves as an effective "pop" screen. On a particularly windy day when a wind screen may not solve the noise problem, or with low frequency noise problems, a hi-pass filter with a cut-off point just above the noise frequency is recommended. For example, in a windy situation where the wind screen is not fully effective, a 300-cycle hi-pass filter will reduce the noise without seriously marring the voice reproduction. Such filters are available in the new Altec line of Audio Control products.

MICROPHONE USAGE

In order to match a microphone to a specific application, the sound engineer should know the capabilities of the various types. Figure 19 is a table indicating the characteristics and pickup patterns for five types of microphones. Across the top are shown the polar curves of each type. These patterns should be regarded as three dimensional. The polar characteristics of the omnidirectional microphone is not a circle, but a sphere; that of the bidirectional resembles two spheres, one 180 degrees from the other at the back of the microphone. The pattern of the cardioid is like a misshapen balloon or, perhaps, a pumpkin with the microphone located at the stem position. The same three-dimensional imagery should be kept in mind for other types, also.

To simplify the voltage output formulas (line 2 of Figure 19), regard that of the omnidirectional microphone as the reference with the directional microphones as cosine functions of that volt-

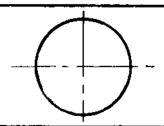
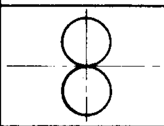
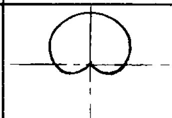
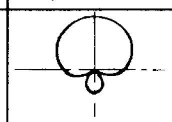
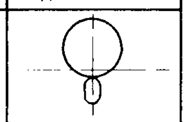
MICROPHONE	Omnidirectional	Bidirectional	Directional	'Super-Cardioid'	'Hyper-Cardioid'
Directional Response Characteristic					
Voltage Output	$E = E_0$	$E = E_0 \cos\theta$	$E = \frac{E_0}{2}(1 + \cos\theta)$	$E = \frac{E_0}{2}\{(\sqrt{3}-1)+(3-\sqrt{3})\cos\theta\}$	$E = \frac{E_0}{4}(1 + 3\cos\theta)$
Random Energy Efficiency (%)	100	33	33	27	25
Front Response / Back Response	1	1	∞	3.8	2
Front Random Response / Total Random Response	0.5	0.5	0.67	0.93	0.87
Front Random Response / Back Random Response	1	1	7	14	7
Equivalent Distance	1	1.7	1.7	1.9	2
Pick-Up Angle (2 θ) For 3 db Attenuation		90°	130°	116°	100°
Pick-Up Angle (2 θ) For 6 db Attenuation		120°	180°	156°	140°

FIGURE 19 - Performance Characteristics of Various Microphones

age. Use of the omnidirectional as the reference for other characteristics shown will simplify understanding of the figure.

Careful study of Figure 19 will show that no one characteristic confers a status of superior merit on any one microphone. The importance of each characteristic is dependent upon the application of the microphone. What is an advantage in one instance becomes a disadvantage under other circumstances.

Reverberation usually is the most important factor in controlling the acoustics of any room. Excessive reverberation causes a piling up or overlapping of successive syllables and results in loss of intelligibility. When sound is reproduced in a live room, the sound pattern is indistinct because the microphone represents only one ear and registers all sounds -- wanted and unwanted.

As the distance between the sound source (speaker) and the microphone increases, syllable articulation decreases. Often this may be overcome by maintaining a short distance between the sound and the microphone so that most of the direct sound is reproduced. In some cases, however, this may result in a loss of presence since it is basically the ratio of the reflected sound to the direct sound which lends an impression of liveness to a recorded or reproduced sound pattern.

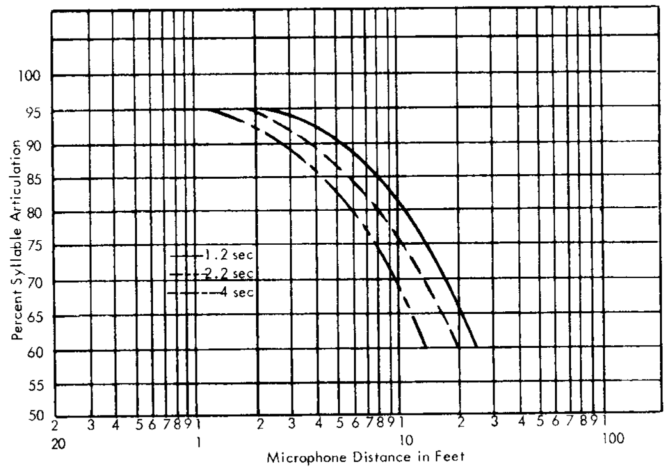


FIGURE 20 - Reduction of Percent Syllable Articulation With Microphone Distance for a Room Having Adjustable Reverberation Time

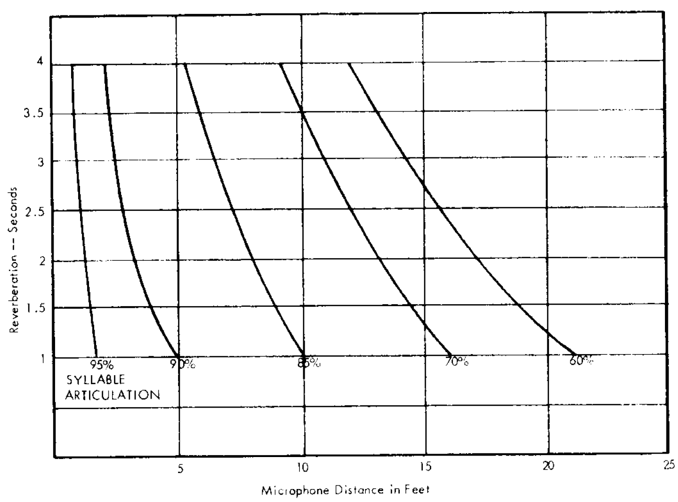


FIGURE 21 - Equiarticulation Contours (from Figure 20)

The reverberation effect upon syllable articulation is shown in Figure 20. Microphone distance is plotted against syllable articulation percentage in a room 20 by 30 by 15 feet. With a 1.2 second reverberation time, the distance would have to be less than two feet in order to obtain an acceptable articulation percentage of 95 percent. At ten feet, the articulation percentage drops to about 80 percent.

The curve in Figure 21, based on the curve in Figure 20, shows the articulation contours plotted against microphone distances with the reverberation time in seconds. As the figure indicates, the microphone distance must be five feet or less in order to attain 90 percent articulation.

MICROPHONE PLACEMENT

Although there is a set of rules regarding the placement of microphones, unfortunately, they do not always work. In some cases it requires a little initiative and a lot of patience on the part of the sound engineer to choose the right microphone for a particular job, and then to decide the correct placement for that microphone.

In an auditorium or room which is not overly reverberant, or where reverberation is not a problem, there are many possibilities. If the speaker is sitting behind a table or otherwise remaining in one location, then a first-class omnidirectional microphone -- be it moving coil or condenser -- would be effective. If, on the other hand, the speaker is moving about somewhat, the choice would be a lavalier-type microphone. A good dynamic lavalier will do an excellent job in 90 percent of such cases. The relatively short distance between the microphone and the sound source provides a large ratio of direct to reflected sounds, and a high order of discrimination against noise and other undesirable sounds. If the speaker is the nervous or the fidgety type who tends to gesticulate and flay his arms around, then the lavalier microphone may pick up noise caused by his clothing brushing against it or caused by "cable-fingering"; problems which are difficult to anticipate. Use of a clip (as furnished with the Altec 686A) will reduce microphone movement against clothing.

The information contained in Figure 19 shows that a cardioid or ribbon microphone may be used at a distance approximately 1.7 times that of an omnidirectional under the same reverberant conditions. In the case of a lavalier-type microphone, this may be an advantage since it is closer to the lips of the speaker (average distance: about ten inches). On the other hand, a cardioid or ribbon microphone would do the same job from a distance of 17 inches (1.7 x 10). If this is an impractical distance, then the lavalier would remain the best choice for a speaker who is moving about.

The response of a pressure-type unit falls off in the higher frequencies when the angle of incidence increases away from zero (perpendicular) incidence. Therefore, a lavalier should have an accentuated high frequency response since it is usually positioned at a 90 degree incidence when it is strapped against the speaker's chest. If the speaker turns his head from side to side, it will cause a further reduction in the high frequency response as well as causing a loss in level. This makes the tilted response characteristic mandatory in a lavalier microphone.

If the head is turned 90 degrees from the front, there will be a reduction of approximately 2 decibels in the output level; an inherent difficulty of using a lavalier-type microphone. By the

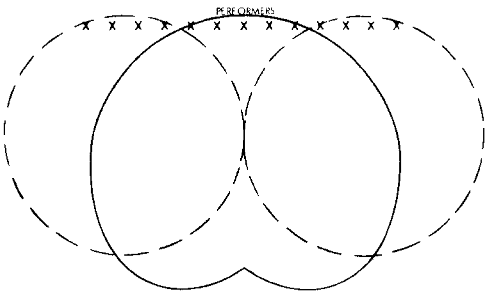


FIGURE 22 - Directional Microphone Used in Place of Two Non-Directional Microphones

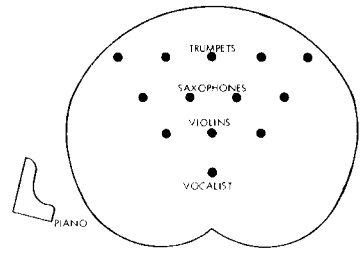


FIGURE 23 - Piano Accompaniment For a Soloist is Picked-Up at a Reduced Sound Level

same token, if the speaker's head is tilted down by 45 degrees, for example, there will be an increase in the sound pressure level of approximately 3 decibels. The same situation is true, to a lesser extend, with any microphone which is used from a distance; but this problem of output level variations is one of the chief difficulties of using the lavalier microphone.

Where there are two speakers, they may be placed on either side of a ribbon microphone with the "dead" side turned toward the audience and the loudspeakers. This will provide a maximum output with a minimum of feedback. A second choice for such an arrangement would be a cardioid microphone placed a little further back than an omnidirectional microphone would be. This will provide broad coverage with a minimum of feedback problems.

Another situation might involve four speakers gathered around a table. Two ribbon microphones, cross-polarized, could be used, but such an arrangement might prove unwieldy. Keeping in mind that the directional characteristics of all microphones are three-dimensional, a cardioid microphone could be used. Placed in a vertical position, a cardioid becomes an omnidirectional microphone. It could be suspended above the speakers or mounted vertically on a stand in the center of the table.

Where feedback and ambient noise do not present a problem, the omnidirectional microphone is good for general purpose use. It is widely used for music and opera broadcasts. For public address systems use other than musical programs, the trend is to replace it with the cardioid. However, for recording and broadcasting purposes (particularly indoors) the omnidirectional still serves as the most useful type of microphone.

Figures 22 through 25 illustrate other typical microphone placement problems. In Figure 22, for example, a dozen or so performers are in a line on a stage; a situation requiring a balanced pickup from the group. The use of two omnidirectional microphones spaced the optimum distance might serve the purpose. However, use of one cardioid might be better inasmuch as the cardioid could be used at a distance 1.7 times further away than the omnidirectional and, as a result, will obtain a better balanced pickup (assuming identical reverberation conditions).

Another pickup problem might involve a soloist accompanied by a piano in which the piano should be slightly subdued. With a simple cardioid placed as indicated in Figure 23, the soloist is picked up while the piano, slightly outside the polar pattern, will produce the required subdued level.

When two performers are spaced some distance apart (as in Figure 24), the cardioid microphone would be better than a pair of omnidirectionals because it may be placed further away. The wide pickup pattern of a single omnidirectional might prevent it from being used.

Reverberation problems could occur in a situation where a public address system is required to cover a small dance band with only one microphone channel; therefore, a cardioid microphone would be the best choice. The proper placement of the various instruments relative to the pickup pattern (proper balance) may be obtained by placing the piano slightly out of the pickup pattern to obtain the reduced level needed. (The same approach would apply to a set of drums if they are used.) The vocalist should be directly on axis and at a relatively short distance from the microphone (Figure 25) for optimum results.

The illustrations given here are only samples of what a sound engineer will encounter. As with all problems, experimentation is the only way to ascertain the best possible solution.

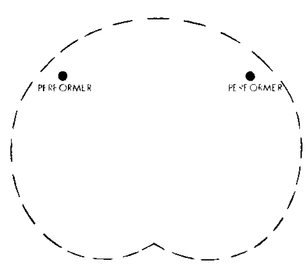


FIGURE 24 - The Wide Pick-Up Angle of a Directional Microphone Requires Little "Panning" for Faithful Pick-Up

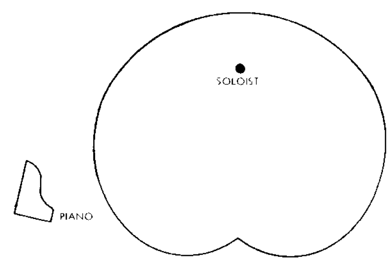


FIGURE 25 - Use of the Directional Microphone for a Small Dance Band

The author wishes to thank Chemical Publishing Co.; 212 Fifth Avenue; New York, N.Y.; for permission to adapt Figures 19 through 25 from material appearing in PRACTICAL ELECTRO-ACOUSTICS (1955) by Michael Rettinger.

OPERATIONAL TECHNIQUES IN PROFESSIONAL RECORDING STUDIOS

H. T. Putnam

President, United Recording Co.

(from a paper presented at the New York A.E.S. Convention, 1962)

INTRODUCTION

The quality of the sound of the final recording is the result of properly controlling many factors. In addition to the elements concerning live recording, other equally important factors such as re-recording, transferring from tape to disc, and disc mastering necessarily enter into the production of a satisfactory final product. This discussion, however, is limited to those factors primarily concerned with the live recording area...namely the recording studio and its associated devices.

Pertaining to live recording then, we are most interested in the following:

1. The recording studio, its physical size and geometry and acoustical characteristics.
2. The control console -- sound mixing devices and associated equipment.
3. Selection of microphones and their performance characteristics.
4. Selection and utilization of reverberation devices.
5. Monitoring systems and their performance.
6. Placement of microphones and physical set-up of the orchestra within the studio.
7. The degree of facility with which the sound mixer technician properly interprets, controls and produces the final desired sound picture.

THE RECORDING STUDIO

Various factors that enter into the performance of the recording studio are as follows:

1. Physical size.
2. Relationship of height, width, and length.
3. The degree of diffusion which exists as a function of the geometry of the studio.
4. Degree of diffusion which exists as a function of the distribution of the acoustical absorbing elements around the studio.
5. Degree of isolation from outside interfering sounds.
6. Relationship of reverberation time to frequency.

With regard to physical size, a chronic problem exists. With rare exception, most commercial recording facilities, because of a combination of economic and physical limitations, find it necessary to record a larger number of performers in the given studio than for which the studio was designed. Theoretical optimums in terms of the relationship between the volume and the number of performers for a given studio are rarely adhered to in practice. By way of comparison, it is interesting to note that, based on the average of theoretical information, a studio with a length of 50 feet, width of 30 feet, and ceiling height of 18 feed, and a total volume of 27,000 cubic feet, would (on an optimum capacity basis) be sufficient for recording a total of fifteen performers. However, in actual practice this studio would be called upon to record over twice that number; up to thirty or thirty-five performers. It is obvious as a result of putting more people in the studio that it is necessary for the various sections of the orchestra to be placed closer together. In order to maintain the necessary and desirable control of the separation, we will resort to a much more close miking technique than we would under ideal circumstances. It has become a most common practice in the production of commercial phonograph records and is the only practical means by which we can control the ratio of direct to indirect or reflected sound, and maintain the necessary sound quality and proper musical perspective.

It is obvious that if the studio were designed to accommodate a lesser number of performers than for which it is used, the reverberation time in the room is reduced as a result of this increased occupancy to a level substantially lower than optimum and that some other means of restoring the desirable reverberation characteristics to the studio is required. This is commonly accomplished by additional reverberation in the respective microphone channels. There are many cases in which the geometric ratios of the studio are far from ideal in terms of height, width and length. It is impossible, however, to ignore the phenomenon of such extreme ratios as 1-2-4, for example. There are certain methods by which the adverse effect of these critical ratios can be minimized if structural changes are impossible.

One practical solution is to arrange the orchestra in the manner that confines it to the area of approximately 50 percent of the total length of the studio. Locating the concentration of the performers at one end or the other, and then by the utilization of highly directional mikes with their back or "dead side" facing the unused portion of the studio, this effect of the undesirable geometry may be minimized to some degree.

It is normally the practice of recording engineers to use a certain number of absorbing flats or screens which are usually arranged in a discreet manner to afford control of the reflected sound that reaches the mikes and could cause interference to the unwanted or direct sound. In the case of studios having these undesirable ratios, large screens and flats may be used to in effect shorten the length of the studio. It may also be necessary to effectively reduce the width of the studio in order to upset the undesirable ratio of width to ceiling height. A number of splays and absorbing screens may be distributed along one wall of the studio at random angles and if the screens and splays cover a sufficient area, the adverse affect of the critical ratio may again be controlled to a satisfactory degree.

Further reduction may be accomplished through using closer mike techniques than would normally be ideal. Diffusion of sound within the recording studio is accomplished by splaying the walls and ceiling over random angles and areas, and by scattering various acoustical elements throughout the walls and ceiling of the studio. The characteristics of a room such as this makes mike placement less critical and lends itself to considerably greater flexibility in terms of orchestral location and set-up.

A choice of optimum reverberation time in designing the recording studio has been a subject of widely diverse opinion. One theory in this connection is directed toward providing better separation and isolation of the various sections of the orchestra lead to a design of a recording facility in which the studio reverberation time was held to a value far below that considered optimum in the past. The degree of separation which was hoped for was achieved. However, a practical problem developed with respect to the fact that the performers felt it extremely difficult to play in this environment. They had the feeling that they must play unnecessarily loud to overcome this condition due to the lack of resonance of sound from their instruments. In this particular situation, the validity of these complaints was of such importance that the studio was later redesigned and provided for more normal reverberation time characteristic commensurate with that necessary to obtain a satisfactory performance from the musicians.

As we pointed out before, most studios are consistently overloaded in terms of optimum capacity and it might seem that the absorption by virtue of increased occupancy would tend to offset some of the problems of separation. This may be true to a very slight degree; however, the location of the performers is dictated by the requirements of the orchestra, and the added amount of absorption is not scattered around the six planes of the room in a manner which would have sufficient effect necessary to offset the separation loss.

The reverberation time as a function of frequency is an important consideration, and in the case of a studio where an insufficient amount of low frequency absorption is present, the room may have a so-called dead sound at middle and high frequencies but produce inadequate separation at low frequencies and cause a problem of recording bass instruments, particularly string bass, properly. This results in a very muddy-sounding finished record, causing the mixing engineer extreme difficulty. This condition is common in some of the older designs in recording and broadcasting studios, and can be solved only by adding a sufficient degree of low frequency absorbing elements in the studio to eliminate the high reverberation time in this portion of the audio spectrum. Some devices commonly used to correct this condition are of the diaphragmatic type such as polycylindrical diffusers, light weight plywood splays, and low frequency absorbing elements such as high density acoustical blankets.

In order to facilitate the greatest utility from the recording studio, the following characteristics should be carefully controlled.

The reverberation time should not be reduced to a level that would inhibit the performance of the musicians or vocalist in the studio.

The reverberation time should not be so high as to make it impossible to obtain satisfactory separation even with the use of close miking techniques.

The placement of the absorbing elements within the studio should be scattered as much as possible.

The concentration of the high amount of absorption in one area is to be avoided.

In order to make the room diffuse to minimize the effect of critical mike placement, the walls and ceiling should be splayed and broken up.

The shortest dimension of these splays should be sufficiently long to affect low frequencies.

The reverberation time versus frequency characteristics of the room should be relatively flat (between 300 to 400 cps.)

The increase of low frequency reverberation time should be limited to approximately 25 percent.

Effective means of varying the reverberation time in the studio to accommodate larger variations in occupancy is desirable.

Geometric ratios which result in a product of whole numbers should be avoided.

SOUND MIXING DEVICES

The flexibility of the control console sound mixing system and its associated equipment determine the degree with which the sound mixing technician can control and record the sounds picked up by the mikes in the studio. The average recording console necessary to meet the requirements of today's recording means should include the following features:

There should be 12 to 16 mike input positions.

Each mike input position should be capable of equalization in order of plus or minus 6 db of boost or attenuation at the low frequencies, and 6 to 10 db of boost at the middle and high frequencies (usually selected at 3000, 5000, or 10,000 cycles).

Each mike channel would further normally incorporate some means of varying the echo send in each channel in order that discreet control of the reverberation on an individual channel basis can be maintained.

Each mike channel usually is provided with facilities to permit switching to one of three (possibly four) program channels at the same time the switching of the echo send busses is accomplished.

Automatic gain reduction facilities are sometimes provided as needed since in many cases it is necessary and convenient to have the means of limiting the peak signals on a particular channel. The use of limiting or automatic gain reduction improves the facility and gives the sound mixer another hand in effect with which to more critically and artistically control the remaining channels.

The use of equalization in addition to the occasional use of low and high pass filters many times facilitates a greater separation since it is possible to shape the characteristic of the mike to favor the sound being picked up on that particular mike, and emphasize the pleasing characteristic of that particular instrument or section of the orchestra. At the same time some of the unwanted sounds which lie outside the range of that particular section may be eliminated.

Many new devices have been developed to facilitate greater convenience and flexibility of the studio mixing system. Some of these are: Manual and automatic gain control devices using light dependent resistors, modular microphone amplifiers incorporating variable equalization, completely transistorized preamplifiers and program amplifiers featuring noise and distortion characteristics comparing favorably with vacuum tubes, controlled threshold devices which effectively cutoff a channel when no program is present thus improving apparent separation.

SELECTION OF MICROPHONES

The wide variety of quality microphones available today creates latitude for selection. However, by and large, personal preference and one's past experience and habits usually become the deciding factors.

It is interesting to note that certain older types such as the Altec 639A and 639B are still in preference in many studios.

Generally speaking, the nature of commercial studio recording is such that more frequent use is made of unidirectional and bidirectional mikes than of omnidirectional types. This reverts to the problem of maintaining control of separation in overloaded (performers) studios as discussed previously.

Another factor which creates a pattern of acceptability and stimulates more universal use of certain types is that some particular microphones favor certain instruments or orchestral sections either by virtue of their response or simply because their polar pattern happens to uniquely adapt itself to a certain application.

One basis of performance evaluation which is frequently ignored regards the quality of the off-axis sound. Since we are not recording in anechoic rooms, there is always a substantial amount of random reflected sound present in the microphone output. It is important that the sound produced by the off-axis pick-up has the audibility which will not degrade the overall sound mixture from the studio.

The front to back ratio of some so-called "cardioid" type microphones is so poor at low frequencies that microphones located remotely from the bass may produce such a level of low frequency pick-up as to destroy the desired presence of the intended pick-up from the bass microphone. Some of the newer type of condenser microphones, however, are excellent in respect to controlling these problems.

Pressure microphones of the dynamic moving coil type and some condenser mikes are bad performers when placed in, or very near, the corner of a studio. This undesirable characteristic manifests itself in the form of exaggerated bass, a general muddy quality resulting from unwanted pick-up of reflected sounds which interfer with the desired program.

Unfortunately, the basis of subjective evaluation and the comparison of microphones places too much emphasis on the aspect of difference in output level of various microphones. A more realistic consideration for comparison would be to compare the microphone quality at equal output levels, subjectively evaluate its "on axis" sound quality, its "off axis" sound quality, its front to back ratio, particularly at low frequencies, rather than just being influenced by the fact that the sound seems to jump out at you, as compared to the lower level microphone, which in most cases is just the result of the increased output and does not necessarily indicate an improved quality.

REVERBERATION DEVICES

Reverberation devices most commonly used in commercial recording are as follows: Acoustical chambers with a total enclosed volume from 1000 to 5000 cubic feet with reverberation times from 1 second to 6 seconds. These rooms are ideally those of good geometric ratios, non-parallel surfaces, and are sufficiently isolated to prevent interference of outside sounds. The driving unit usually is a high quality loudspeaker (or speakers) operated at a nominal power level of 2 or 3 watts, with the acoustic output attenuated below 200 or 250 cycles at 6 db per octave or more, by either electrical or acoustical means, High quality microphones usually a cosine or omnidirectional polar pattern are employed. Frequently two channels are fed into two separate speakers and picked up by two individual microphones. Partitions or screens are sometimes employed to further separate the direct sound.

Satisfactory results may be obtained for commerical two-channel stereo records. The microphones and speakers are physically arranged so that the arrival time is sufficiently different from the left speaker to the left microphone as compared to the right microphone that the precedence effect is maintained sufficiently to create an acceptable degree of separation.

The electro-mechanical type which has been given the widest acceptance has been the driven steel plate such as the EMT. Recently a two-channel version of this unit has become available.

The other electro-mechanical types such as the spring-driven devices have not found particularly wide acceptance in the professional field, except for special effects use.

Tape reverberation is sometimes employed for special effects and may be used in some instances to delay the echo send feed or receive form an acoustic or electro-mechanical chamber for producing a certain echo perspective.

Consistently the objective is to add reverberation which does not adversely "color" the sound. This requires that, ideally, the reverberation channel has a flat frequency characteristic. This is difficult to achieve. A smooth decay characteristic of the chamber or device is also an important factor. Equalization is commonly used in the send or receiver to compensate for some of the frequency irregularities, as well as for producing accented effects in certain ranges when desired. A response characteristic measured by selected band widths of white noise indicates a frequency response of a typical acoustical chamber to be plus or minus 10 db from 200 to 6000 cps (6 kc).

A new reverberation unit incorporating an electrostatic recording method will shortly be commercially available. Preliminary results indicate adaptability to the professional field.

MONITOR SYSTEMS

Monitoring systems in professional recording are perhaps less standardized then facilities of any other single area. Many diverse opinions exist, such as these:

1. We should monitor so that the acoustic level in the control room is exactly equal to the acoustic level in the studio.

2. We should monitor on a system comparable to the quality and range of a good home music system at typical "living room" volume levels.

3. We should monitor at a level in excess of the live studio level in order to be able to hear more critically.

4. We should monitor on a high quality uncompensated professional studio type system at a level slightly above that of "living room" level.

In professional studio practice, the average operating level probably falls between condition 1 and condition 4. Sound mixers for the most part fall into a pattern as a matter of personal preference and habit. Certain other factors which affect the operating level have to do with fatigue. This causes a trend of constantly increasing the level as the recording session progresses. By actual observation, a consistent pattern of increasing monitor levels of 6 db to 10 db from the start to the finish of a three-hour recording session is typical.

Acoustic levels of 105 db are frequently reached in the control booth.

Probably the most universally used speaker system in professional studios is the Altec 604 D, now replaced by the 605A*, and the Altec A-7 systems where space is adequate. Due to the power levels required under some of the previous conditions, 50 to 60 watt high quality amplifiers are commonly used. Perhaps the only concerted statement on which there would be little or no disagreement is that there is an urgent need for standardization and for vast improvement in the field of professional studio monitoring loudspeakers.

---

*Recently Altec revived the 604 series with the appearance of the new 604 E model.

MICROPHONE PLACEMENT AND PHYSICAL PLACEMENT OF PERFORMERS IN THE RECORDING STUDIO

Before it is possible to select microphones and locations, we must first satisfy certain requirements of the physical set-up of the performers or orchestra within the studio. A very important function of the sound mixer is the planning of the arrangement of the orchestra in the studio. This is sometimes done with the consultation of the arranger or conductor and the A & R (artist and repetoire) man. It is virtually impossible in most cases to make major changes in the orchestra set-up during the recording session period without accruing a tremendous loss of time and increased cost. This emphasizes the need for great skill and careful analysis of the unique requirements of a particular orchestral set-up. The problem is further compounded by the demands for producing certain stereo effects which may dictate some aspects of the physical set-up. Each combination of performers or orchestras have unique needs which relate to the style or type of music to be recorded, but certain basic demands must be satisfied.

For pruposes of discussion, let's consider a typical "pop" orchestral set-up and vocalist consisting of the following instrumentation:

Solo vocal
Vocal group (8 people)
Four rhythm (piano, bass, guitar, drums)
Five woodwinds
Eight brass (4 trumpets, 4 trombones)
Three French horns
Sixteen violins
Four viola
Four celli
Vibes, typani, xylophone, chimes, misc. percussion
Solo electric guitar
Harp
Celeste

Some of the primary considerations of the physical orchestral set-up are:

1. The rhythm section must be relatively tightly grouped.

2. The utility percussion players and solo guitar may be playing rhythm part of the time and must, therefore, be fairly close to the rhythm section.

3. The solo vocalist as well as the vocal group must have good rapport with the rhythm section which dictates their location.

4. The French horns are frequently concerted in harmony or in unity with the brass and must be located fairly close to the brass section.

5. The brass section must be located so that it is anchored by the rhythm section, particularly the drummer.

6. The string section consisting of violins, violas and celli must be able to hear each other as well as the rhythm, but must be located with a view toward maintaining maximum control of separation from the brass.

7. The woodwinds must have equally good rapport with the rhythm section.

There are many other supplemental considerations but aside from the physical requirements of the orchestra, the other big concern is maintaining the necessary separation to produce a product of acceptable musical and sound quality.

The arranger for the commercial pop record field has been led to expect miracles from the sound technicians simply because it has been and is being done, through the ingenious combination of outstanding operational performance by the sound mixer and fine electronic apparatus.

For example, let's consider the intensity levels involved in the various sections of the orchestra. The difference in maximum acoustic output level from the total string section is more than 20 db less than that of the eight brass; yet, contrary to the situation of the concert orchestra which is basically musically balanced within itself. The commerical record orchestra must on some occasions, by design of the arranger, create a musical picture that requires the string section to be equally loud to the brass. Let's see what happens if certain precautions were not taken. Assume the brass were located to be picked up on the right channel microphone, and the string section on the left. As we increase the gain to produce the necessary string section level, we may find the brass section suddenly losing presence, and in effect, being picked up more on the string mike, or left channel. As the producer calls for more brass, he has actually reacted to the change in sound quality due to the "off mike" pick-up. There isn't really any change in the intensity of the brass channel, which is still set at the same original gain level. To minimize this effect, we may do the following:

1. Reduce the distance of the string section microphone pick-up, as we do this the number of instruments within the on-axis field is reduced, therefore more microphones are required. (A 50 percent reduction in distance reduces the unwanted off-axis sound ratio by 6 db.)

2. Establish a location of the string section with respect to brass to eliminate as much direct sound as possible from the brass section. This would mean to certainly avoid a location wherein the on-axis violin mike would pick up direct brass sound.

3. Utilize mikes with extremely good front to back ratio and whose off-axis response does not adversely affect sound quality.

4. Locate absorbing screens behind the section in line with the on-axis pattern to eliminate random pick-up from reflecting surfaces behind the section.

5. Artistically dump the gain of the string channel to the minimum level required at any given instant, to satisfy the desired musical perspective.

The foregoing is only one isolated example of some of the problems the sound mixer faces. It is truly amazing that the outstanding results which are evidenced by the quality of many commercial records today can be attained under circumstances which are so contradictory to ideal conditions.

The qualifications of a successful sound mixer represent a unique and rare combination of skills. He must be adequately technically oriented to understand and evaluate the performance of the various electronic and acoustical devices with which he works. He must have sufficient musical aptitude to interpret the wishes of the arranger-conductor; he must be creatively artistic, imaginative, have a flair for showmanship, willing to try the impossible, and have the ambidexterity of an octopus. He must have the unique talent of being able to communicate with artists and directors at any artistic level, be able to perform his functions deftly under extreme pressure, and, above all, must have the patience of Job.