AUDIO APPLICATIONS FOR MEETING FACILITIES

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Introduction

Audio for meeting facilities is a topic that encompasses a wide range of applications, from a simple public address system in a courtroom to a multi-channel legislative system with recording and broadcast capabilities. Though these systems may vary in size and complexity, they are all governed by the same physical principles and they share certain types of equipment.

Common components of these systems include microphones, automatic and non-automatic mixers, power amplifiers, loudspeakers, and many types of electronic signal processing devices such as equalizers, compressors, and audio time delays. A complete audio system may involve some or all of these items. Proper selection and application of this equipment requires knowledge of both the intended purpose of the sound system and the characteristics of individual components.

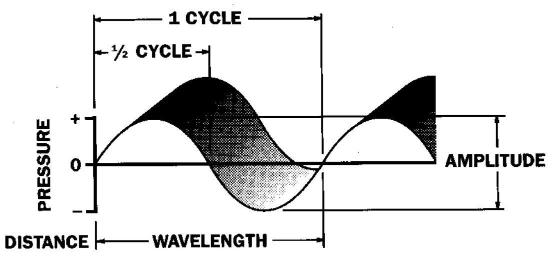
The scope of this guide is limited to the selection and application of wired microphones, wireless microphones, and microphone mixers for meeting facility sound systems. Since microphones and mixers act as the interface between the sound source (the talker) and the sound system, it is imperative to thoroughly discuss these two subjects, as well as sound in general. The objective is to provide the reader with sufficient information to understand how microphones and mixers are applied to meeting facility sound situations.

Sound

Because "good" sound quality is the goal of any meeting facility sound system, it is helpful to be familiar with some general aspects of sound: how it is produced, transmitted, and received. In addition, it is also useful to describe or classify sound according to its acoustic behavior.

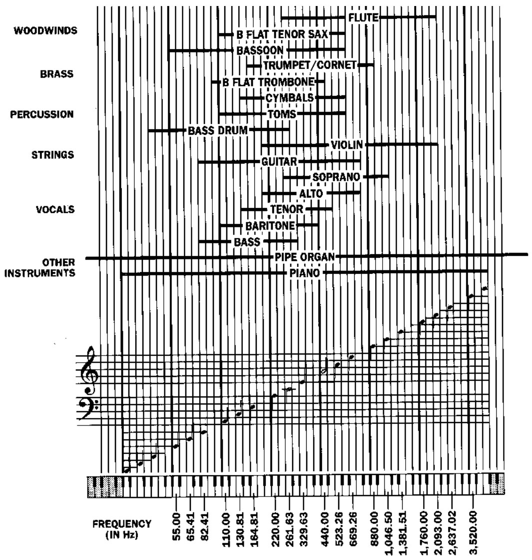
Sound is produced by vibrating objects. These include musical instruments, loudspeakers, and, of course, human "vocal cords". The mechanical vibrations of these objects move the air which is immediately adjacent to them, alternately "pushing" and "pulling" the air from its resting state. Each back-and-forth vibration produces a corresponding pressure increase (compression) and pressure decrease (rarefaction) in the air. A complete pressure change, or "cycle", occurs when the air pressure goes from rest (or "normal"), to maximum, to minimum, and back to rest again. These cyclic pressure changes travel outward from the vibrating object, forming a pattern called a sound wave. A sound wave is a series of pressure changes (cycles) moving through the air.

A sound wave can be described by its frequency and by its amplitude. The frequency of a sound wave is the rate at which the pressure changes occur. It is measured in Hertz (Hz), where 1 Hz is equal to 1 cycle-per-second. The range of human hearing extends from a low of about 20 Hz to a high of about 20,000 Hz. In practice, a sound source usually produces many frequencies simultaneously.



Elements of a sound wave

The amplitude of a sound wave refers to the magnitude (strength) of the pressure changes and determines the "loudness" of the sound. Amplitude, or sound pressure level (SPL), is measured in decibels (dB) and ranges from 0 dB SPL (the threshold of hearing), to above 120 dB SPL (the threshold of pain). The level of conversational speech is about 65 dB SPL. A change of 1 dB is about the smallest SPL difference that the human ear can detect, while 3 dB is a noticeable step, and an increase of 10 dB is perceived as a "doubling" of loudness.



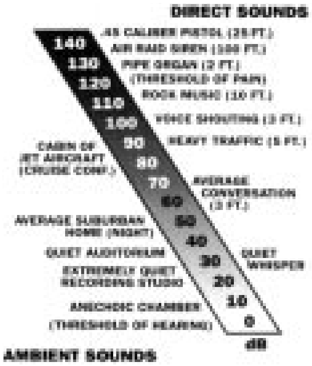
Fundamental frequencies of sound sources

Another characteristic, related to frequency, is wavelength. The wavelength of a sound wave is the physical distance from the start of one cycle to the start of the next cycle, as the wave moves through the air. Since each cycle is the same, the distance from any point in one cycle to the same point in the next cycle is also one wavelength: for example, the distance from one maximum pressure point to the next maximum pressure point. Wavelength is related to frequency by the speed of sound.

The speed of sound is the velocity at which a sound wave travels. The speed of sound is constant and in air is equal to about 1130 feet per second. It does not change with frequency or wavelength, but it is related in the following way: the frequency of a sound, multiplied by its wavelength, always equals the speed of sound. Thus, the higher the frequency of sound, the shorter the wavelength, and the lower the frequency, the longer the wavelength. A 20 to 20,000 Hz frequency range corresponds to a maximum wavelength of about 55 feet. at 20 Hz to a minimum wavelength of about one-half inch at 20,000 Hz. This large range of wavelengths is responsible for many acoustic effects, both desirable and undesirable.

Sound is transmitted through some “medium”. Air is the typical medium, but sound can also be transmitted through solid or liquid materials. Generally, a sound wave will move in a straight line unless it is absorbed or reflected by physical surfaces or objects in its path. However, the sound wave will be affected only if the surface is large compared to the wavelength of the sound. If the surface is small (compared to the wavelength) the sound will proceed as if the object were not there. High frequencies/short wavelengths can be reflected or absorbed by small surfaces, but low frequencies/long wavelengths can only be reflected or absorbed by very massive surfaces or objects.

Once a sound has been produced and transmitted, it can be received by the ear and, of course, by a microphone. In the ear, the arriving pressure changes “push and pull” on the eardrum. The resulting motion of the eardrum is converted by the inner ear into nerve signals that are ultimately perceived by the brain as “sound”. In a microphone, the pressure changes act on a diaphragm. The resulting diaphragm motion is converted into an electrical signal that is sent to the sound system. The resultant sound picked up is a combination of all pressure changes occurring at the eardrum or microphone diaphragm.

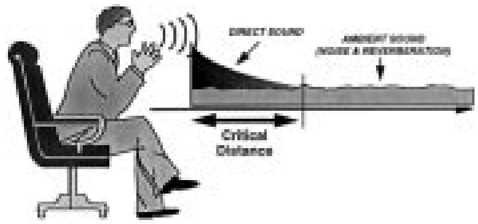


Sound pressure level of typical sound sources

Sound can be classified by its acoustic behavior. An example is “direct” sound versus “ambient” (or indirect) sound. Direct sound travels from the sound source to the listener in a straight line (the shortest path). Indirect sound is reflected by one or more surfaces before reaching the listener (a longer path). Since sound travels at a constant speed, it takes a longer time for the indirect sound to arrive, and is delayed relative to the direct sound. There are several kinds of indirect sound, depending on the room acoustics.

Echo occurs when an indirect sound is delayed long enough (by a distant reflecting surface) to be heard by the listener as a distinct repetition of the direct sound. If indirect sound is reflected many times from different surfaces it becomes “diffuse” or non-directional. This is called reverberation, and it is responsible for our auditory perception of the size of a room. Reverberant sound is a major component of ambient sound, which may include other non-directional sounds, such as wind noise or building vibrations. A certain amount of reverberant sound is desirable to add “depth” and “fullness” to the sound, but an excess tends to make the sound “muddy”, “hollow”, and unintelligible.

One additional form of indirect sound is known as a standing wave. This may occur when the wavelength of a sound is the same distance as some major dimension of a room, such as the distance between two opposite walls. If both surfaces are acoustically reflective, the frequency corresponding to that wavelength will be amplified, by addition of the incoming and outgoing waves. This results in a strong, stationary wave pattern between the two surfaces. A standing wave happens primarily with low frequencies, which have long wavelengths and are not easily absorbed.



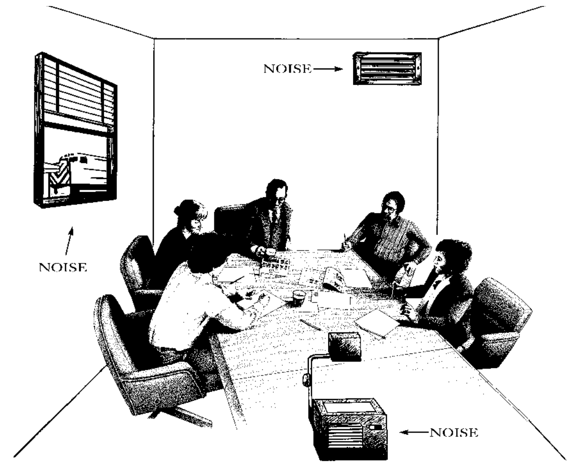
Direct sound decreases with distance and is eventually no louder than ambient sound.

An additional property of direct sound is that it becomes weaker as it travels away from the sound source, at a rate governed by the inverse-square law. For example, when the distance increases by a factor of two (doubles), the sound level decreases by a factor of four (the square of two). This corresponds to a drop of 6 dB in sound pressure level (SPL), which is a substantial decrease. In contrast, ambient sound, such as noise and reverberation, is at a constant level everywhere in the space. Therefore, at a given distance from a sound source, a listener (or a microphone) will pick up a certain proportion of direct sound to ambient sound. As the distance increases, the direct sound level decreases while the ambient sound level stays the same. A properly designed sound system can increase the amount of direct sound reaching the listener without increasing the ambient sound significantly.

The Sound Source

The sound source most often found in meeting facility applications is the speaking voice. Voices may be male or female, loud or soft, single or multiple, close or distant. Pre-recorded audio from video or audio tape is also very common in a meeting facility.

In addition to these desired sound sources there are certain undesired sound sources that may be present: building noise from air conditioning, buzzing light fixtures, noise from meeting participants, sounds from street or air traffic, etc. All these undesired sounds can interfere with the desired sound source. (See figure below.)



Sources of possible acoustical interference in a room—open window, hard walls, overhead projector, ventilation shaft

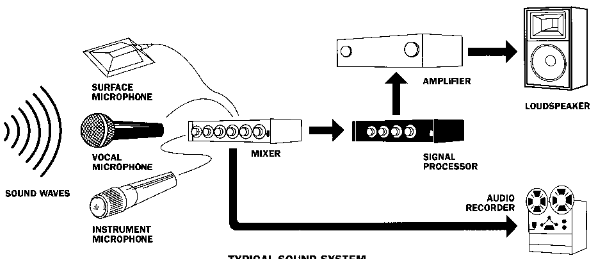
In this context, the loudspeakers of the sound system must also be considered as a sound source. They are a desired sound source for the meeting participants, but an undesired source for microphone pickup. Feedback (an annoying howl or ringing sound) can occur in any sound system if microphones "hear" too much of the loudspeakers.

Finally, the acoustics of the room are often as important as the sound source itself. Room acoustics are a function of the size and shape of the room, the materials covering the interior surfaces, and even the presence of the human bodies which absorb sound. The acoustic nature of an area may have a positive or a negative effect on the sound produced by talkers and loudspeakers before the sound is picked up by microphones or heard by listeners. Room acoustics can absorb and diminish some sounds while reflecting or reinforcing other sounds. The latter can contribute to undesired sound in the form of echo or excessive reverberation. In general, intelligibilty problems caused by room acoustics must be solved by acoustic means, not electronic means.

In review, sound sources may be categorized as desired or undesired, and the sound produced may be further classified as direct or ambient. In practice, the "soundfield" or total sound in a space will always consist of both direct and ambient sound except in scientific "anechoic" chambers or, to some extent, outdoors, when there are no nearby reflective surfaces.

The Sound System

A basic sound reinforcement system consists of an input device (microphone), a control device (mixer), an amplification device (power amplifier), and an output device (loudspeaker). This arrangement of components is sometimes referred to as the audio chain: each device is linked to the next in a specific order. The primary goal of the sound system in meeting facility sound applications is to deliver clear, intelligible speech to each meeting participant. The overall design, and each component of it, must be carefully thought out, properly installed, and thoughtfully operated to accomplish this goal.



TYPICAL SOUND SYSTEM

There are three levels of electrical signals in a sound system: microphone level (a few thousandths of a volt), line level (approximately one volt), and loudspeaker level (ten volts or higher).

Sound waves are converted into an equivalent electrical signal by the microphone. This microphone level signal is amplified to line level and possibly combined with signals from other microphones by the mixer. The power amplifier then boosts the line level signal to loudspeaker level to drive the loudspeakers, which convert the electrical signal back into sound.

Electronic signal processors, such as equalizers, limiters or time delays, are inserted into the audio chain, usually between the mixer and the power amplifier, or often within the mixer itself. These devices operate at line level and their general function is to enhance the sound in some way or to correct certain deficiencies in the sound sources.

In addition to feeding loudspeakers, the output of the system may be sent simultaneously to recording devices or even used for broadcast. It is also possible to deliver sound to multiple rooms by using additional power amplifiers and loudspeakers.

Finally, it is necessary to consider the room acoustics as part of the sound system: acoustics act as a "signal processor" that affects sound both before it is picked up by the microphone and after it is produced by the loudspeakers. Good acoustics may enhance the sound, while poor acoustics may severely degrade it, usually beyond the corrective capabilities of the equipment. In any case, the role of room acoustics in sound system performance cannot be ignored.

Microphones: Characteristics, Selection

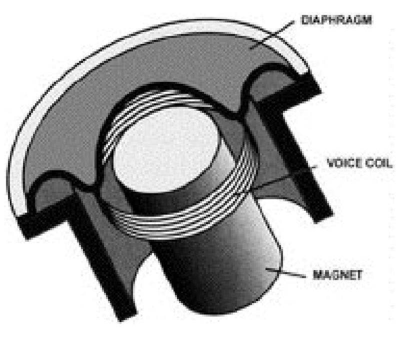
The microphone is the first link in the audio chain and is therefore critical to the overall performance of a sound system. Improper selection of microphones may prevent the rest of the system from functioning to its full potential. Proper selection of microphones depends on an understanding of basic microphone characteristics and on a knowledge of the intended application.

To be most effective, a microphone must be matched both to the desired sound source (a voice, a musical instrument, etc.) and to the sound system (a PA system, a tape recorder, etc.) with which it is used. There are five areas of microphone specifications that must be considered when selecting a microphone for a particular application. These are: 1) the operating principle of the microphone, 2) the frequency response of the microphone, 3) the directionality of the microphone, 4) the electrical output characteristics of the microphone, and 5) the physical design of the microphone.

1) Operating Principle:
How does the microphone change sound into an electrical signal?

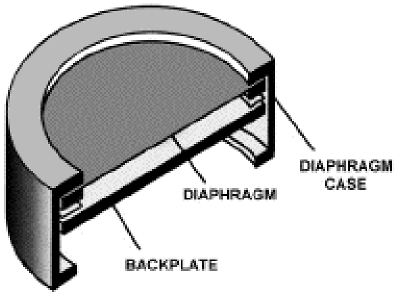
The operating principle describes the kind of transducer inside the microphone. A transducer is a device that changes energy from one form into another, in this case, acoustic energy into electrical energy. It is the part of the microphone that actually senses air movement (sound) and converts it into an electrical signal. The operating principle determines some of the basic capabilities of the microphone.

The two most common types are dynamic and condenser. Although there are other operating principles used in microphones, such as crystal, carbon, etc., these are used primarily in radio communications systems or are only of historical interest. They are almost never encountered in meeting facility sound applications.



Dynamic microphones employ a diaphragm/voice coil/magnet assembly which forms a miniature sound-driven electrical generator. Sound waves travel to the microphone and strike a thin plastic membrane (diaphragm) which vibrates in response. A small coil of wire (voice coil) is attached to the rear of the diaphragm and vibrates with it. The voice coil itself is surrounded by a magnetic field created by a small permanent magnet. It is the motion of the voice coil in this magnetic field which generates the electrical signal corresponding to the sound picked up by a dynamic microphone.

Dynamic microphones have relatively simple construction and therefore are economical and rugged. They can be manufactured with excellent sound quality and with good specifications for use in every area of microphone applications. Dynamics are most widely used in general sound reinforcement and have many uses in meeting facility sound systems.



Condenser microphones are based on an electrically-charged diaphragm/backplate assembly which forms a sound-sensitive capacitor. Here, sound waves vibrate a very thin metal or metal-coated-plastic diaphragm. The diaphragm is mounted just in front of a rigid backplate which is metal or metal-coated-ceramic. In electrical terms, this assembly is known as a capacitor (historically called a condenser) and has the ability to store an electrical charge or voltage. When the condenser element is charged, an electric field is created between the diaphragm and the backplate, proportional to the spacing between them. It is the variation of this spacing, due to the motion of the diaphragm relative to the backplate, that produces the electrical signal corresponding to the sound picked up by a condenser microphone.

The construction of a condenser microphone must include some provision for maintaining the electrical charge. An electret condenser microphone has a permanent charge, maintained by a special material deposited on the backplate or on the diaphragm. Other types are charged by means of an external power source.

All condensers contain additional circuitry to match the electrical output of the element to typical microphone inputs. This requires that all condenser microphones be powered: either by an internal battery or by phantom power, which is a method of supplying power to a microphone through the microphone cable. There are two potential limitations of condenser microphones due to the additional circuitry. First, the electronics produce a small amount of noise. Second, there is a limit to the maximum signal level that the electronics can handle. Good condenser designs, however, have very low noise levels and are also capable of very wide dynamic range.

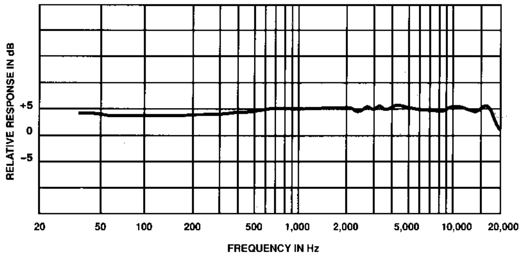
Condenser microphones are more complex than dynamics and tend to be somewhat more costly. However, condensers can readily be manufactured with higher sensitivity and can provide a smoother, more natural sound, particularly at high frequencies. Flat frequency response and extended frequency range are much easier to obtain in a condenser. In addition, condenser microphones can be made very small physically without significant loss of performance.

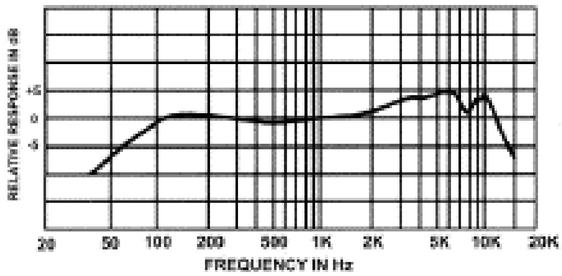
The decision to use a condenser or dynamic microphone depends not only on the sound source and signal destination, but on the physical setting as well. From a practical standpoint, if the microphone will be used in a demanding application such as an audience microphone that is passed around or for outdoor use, a dynamic microphone is the better choice. In a more controlled environment, like a boardroom, auditorium, or courtroom, a condenser microphone might be preferred, especially when the highest sound quality is desired.

2) Frequency Response:
How does the microphone sound?

The frequency response of a microphone is defined by the range of sound (from lowest to highest frequency) that it can reproduce, and by its variation in output within that range. It is the frequency response that determines the basic "sound" of the microphone. The two general types of frequency response are flat and shaped. These terms refer to the graphical representation of frequency response or response curve.

A microphone that provides a uniform output at every audible frequency will have a frequency response graph that is an even, flat line. This is said to be a flat response. This means that the microphone reproduces all of the sound within its frequency range with little or no variation from the original sound. In addition, flat response microphones typically have an extended frequency range, i.e., they can reproduce very high and/or very low frequencies as well. Wide range, flat response microphones have a natural, "uncolored" sound.





By contrast, a microphone with a shaped response will have a frequency response graph with a varying line with specific peaks and dips. This shows that the microphone is more sensitive to certain frequencies than to others, and often has a more limited frequency range. A shaped response is usually designed to enhance the sound of a particular source in a particular application, while at the same time minimizing the pickup of certain unwanted sounds. Shaped response microphones each have a "characteristic" sound.

The selection of a flat or shaped response microphone involves consideration of both the sound source and the sound destination. The frequency range of the microphone must be wide enough to pick up the desired range of the sound source. This range must also be appropriate to the intended destination of the sound: that is, wider range for high quality sound systems or recording/broadcast systems, narrower range for speech-only public address systems.

Within its range the microphone should respond in such a way that the sound is reproduced either with no change (flat response) or with changes that enhance the sound in some desirable manner (shaped response) Normally, microphones with flat, wide range response are most useful for music applications.

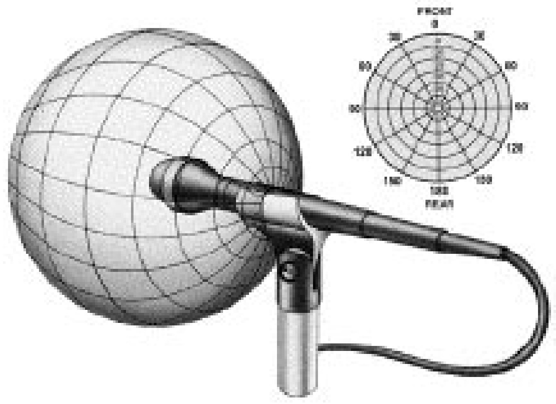
For speech use, the most common response is shaped. Typically, this consists of limiting the range to that of the human voice and adding an upper midrange response rise. Such a presence rise, coupled with controlled low and high frequency response can provide a sound with improved vocal clarity. This is especially true for lapel or lavalier microphones.

Finally, the frequency response of some microphones is switch adjustable to tailor the microphone to different applications. The most common are a low frequency rolloff switch, which can help prevent “rumble”, and a presence rise switch to enhance intelligibility.

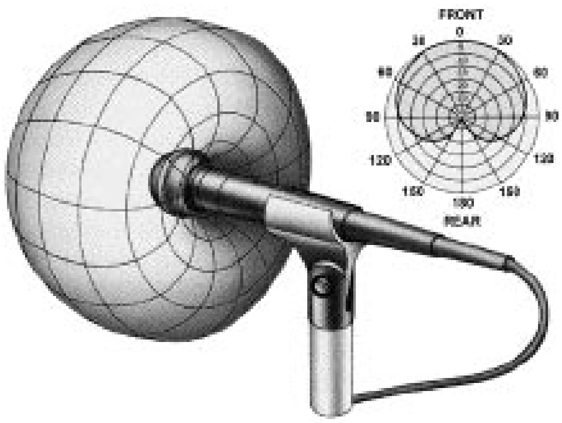
3) Directionality:
How does the microphone respond to sound from different directions?

The directional characteristic of a microphone is defined as the variation of its output when it is oriented at different angles to the direction of the sound source. It determines how best to place the microphone, relative to the sound source(s), to enhance pickup of desired sound, and to minimize pickup of undesired sound. The polar pattern of a microphone is the graphical representation of its directionality. The two most common directional types are omnidirectional and unidirectional.

A microphone that exhibits the same output regardless of its orientation to the sound source will show on a polar graph as a smooth circle and is said to have an omnidirectional pattern. This indicates that the microphone is equally sensitive to sound coming from all directions. An omnidirectional microphone can therefore respond to sound from a wide area, but cannot be “aimed” to favor one sound over another.



A unidirectional microphone, on the other hand, is most sensitive to sound coming from only one direction. On a polar graph, this will appear as a rounded but non-circular figure. The most common type of unidirectional microphone is called a cardioid (like cardiac), because of its heart-shaped polar pattern.



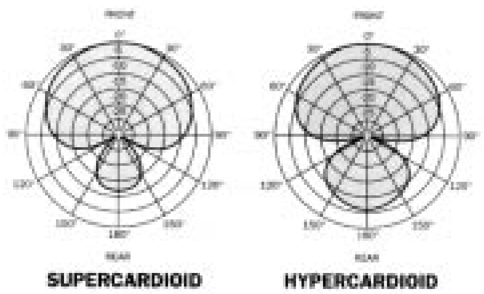
A cardioid type is most sensitive to sound coming from in front of the microphone (the bottom of the heart shape). On the polar graph this is at 0 degrees, or on axis. A cardioid microphone is less sensitive to sound coming from the sides (off axis), and least sensitive to sound from the rear (the notch at the top of the heart shape). For any microphone, the direction of least sensitivity (minimum output) is called the null angle. For a cardioid pattern this is at 180 degrees, or directly behind the microphone.

Thus, a unidirectional microphone may be aimed at the desired sound source by orienting its axis toward the sound. It may also be aimed away from undesired sound sources by orienting its null angle toward these sounds. In addition, a unidirectional microphone picks up less unwanted ambient sound than an omnidirectional, due to its overall lower sensitivity at the sides and rear. For example, a cardioid picks up only one-third as much ambient sound as an omnidirectional type.

Although the output of a unidirectional microphone is maximum for sound arriving at an angle of 0 degrees (on axis), it falls off only slightly for sound arriving from within a certain angle off-axis. The total directional range for usable output is called the coverage angle or pickup arc. For a cardioid microphone, this angle is about 130 degrees.

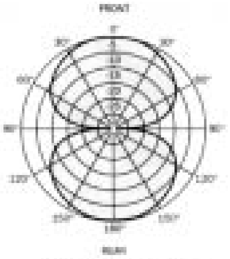
Two related types of unidirectional microphones are the supercardioid and the hypercardioid. Compared to a cardioid type, these have a progressively narrower coverage angle: 115 degrees for a supercardioid and 105 degrees for a hypercardioid. However, unlike the cardioid, they have some pickup directly behind the microphone. This is indicated in their polar patterns by a rounded projection, called a lobe, toward the rear of the microphone. The direction of least sensitivity (null angle) for these microphone types is about 125 degrees for the supercardioid and 110 degrees for the hypercardioid. In general, any directional pattern that has a narrower front coverage angle than a cardioid, will have some rear pickup and a different direction of least pickup.

The significance of these two polar patterns is their greater rejection of ambient sound in favor of on-axis sound: the supercardioid has the maximum ratio of on-axis pickup to ambient pickup, while the hypercardioid has the overall least pickup of ambient sound (only one-quarter as much as an omni). These can be useful types for certain situations, such as more distant pickup or in higher ambient noise levels, but they must be placed more carefully than a cardioid to get best performance. And these microphone types do not necessarily provide better feedback rejection than a cardioid.



Other types of unidirectional microphones include the shotgun and parabolic reflector. These have extremely narrow pickup patterns and are used for distant pickup or in very high ambient noise situations. These specialized microphones are utilized for broadcast and feature film production, not for public address systems. Their sound quality and mechanical limitations make them generally unsuitable for typical meeting facility use.

One additional directional microphone is the bidirectional type. As the name implies, it is equally sensitive to sound from two directions: directly in front of the microphone and directly behind it. Its polar graph consists of a front pickup area and an identical rear lobe, and resembles a figure 8 pattern. Although the front coverage angle of a bidirectional microphone is only 90 degrees, it has equal rear coverage. The null angle is at 90 degrees, which is directly at the side of the microphone. While the bidirectional microphone is not used by itself in any typical meeting facility applications, it is occasionally used in combination with other microphone types for stereo sound recording.



BIDIRECTIONAL

It should be noted that this discussion of directionality assumes that the polar pattern for a microphone is uniform, that is, the same shape at all frequencies. In practice, this is not always achieved. Most microphones maintain their nominal polar pattern over only a limited range of frequencies. This is the reason that published polar patterns include curves measured at several frequencies. High quality, well-designed microphones are distinguished by the uniformity of their polar pattern over a wide frequency range and by the similarity of the pattern to the theoretical ideal.

CHARACTERISTIC	OMNI-DIRECTIONAL	CARDIOID	SUPER-CARDIOID	HYPER-CARDIOID	BI-DIRECTIONAL
POLAR RESPONSE PATTERN					
COVERAGE ANGLE	360°	131°	115°	105°	90°
ANGLE OF MAXIMUM REJECTION (NULL ANGLE)	—	180°	126°	110°	90°
AMBIENT SOUND SENSITIVITY (RELATIVE TO OMNI)	100%	33%	27%	25%	33%
DISTANCE FACTOR (RELATIVE TO OMNI)	1	1.7	1.9	2	1.7

There are a few operational differences between omnidirectional and unidirectional microphones. A useful feature of most unidirectional types is proximity effect. This refers to the increased low frequency response of a unidirectional microphone when it is placed closer than one or two feet to the sound source. Proximity effect becomes most noticeable at very short distances where typically there is a substantial bass boost at less than two inches. Proximity effect adds fullness and warmth to the sound, but can also "muddy" the sound in speech applications. Omnidirectional microphones do not exhibit proximity effect. Omnidirectional microphones also are less sensitive to wind noise and to handling noise. Most professional unidirectional types have effective built-in windscreens and shock mounts to compensate.

Selecting an omnidirectional or unidirectional microphone again depends on the sound source and the destination of the audio signal. For recording (but not sound reinforcement) of meeting participants, an omnidirectional microphone may be used to pick up sound from all directions rather than emphasizing individual voices. However, as part of a sound reinforcement or P.A. system, an omnidirectional microphone may be more prone to feedback because it cannot be aimed away from the loudspeakers.

A unidirectional model can not only help to isolate one talker from other nearby talkers, but can also partially reject background noise. In addition, a unidirectional microphone, properly placed, minimizes feedback, allowing higher sound reinforcement levels. For these reasons, unidirectional microphones outnumber omnidirectional microphones in most meeting facility sound applications.

4) Electrical output:
How does the microphone output match the sound system input?

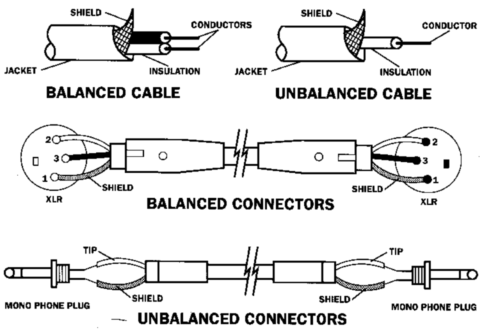
The electrical output of a microphone is characterized by its sensitivity, its impedance, and by its wiring scheme. The same characteristics are used to describe microphone inputs in sound systems. This determines the proper electrical match of a microphone to a given sound system.

The sensitivity of a microphone is defined as its electrical output level for a certain input sound level. The greater the sensitivity, the higher the electrical output will be for the same sound level. The sensitivity should be within a range that will deliver a reasonable signal level to the sound system input: not so high that it will overload the input and not so low that electrical noise is noticeable. It should be noted that for weak or distant sound, a microphone of high sensitivity is desirable, while loud or closeup sound can be picked up well by lower sensitivity microphones.

Impedance is, approximately, the output electrical resistance of the microphone: 150-600 ohms for low impedance (low Z), 10,000 ohms or more for high impedance (high Z). While the majority of microphones fall into one of these two divisions, there are some that have switchable impedance selection. In any case, the choice of impedance is determined by two factors: 1) the length of cable needed to go from the microphone to the mixer input, and 2) the rated impedance of the mixer input.

The maximum length of cable that may be used with a high impedance microphone should be limited to no more than 20 feet. For longer cable lengths, the high frequency response of the microphone will be progressively diminished. Low impedance microphones, on the other hand, may be used with cables of 1000 feet or more with no loss of quality, and are therefore recommended for most meeeting facility applications.

The output wiring scheme of a microphone can be either balanced or unbalanced. A balanced output carries the signal on two conductors. A shield surrounds these the two conductors to keep out unwanted signals. The signals on each conductor are the same level but they are of opposite polarity. A balanced (or differential) input is sensitive only to the difference between the two signals and ignores any part of the signal which is the same in each conductor. Because of the close proximity of the two conductors in a balanced cable, any noise or hum picked up by the cable will be of the same level and the same polarity in each conductor. This common-mode noise will be rejected by the balanced input, while the original microphone signal is unaffected. This greatly reduces potential noise in balanced microphones and cables.

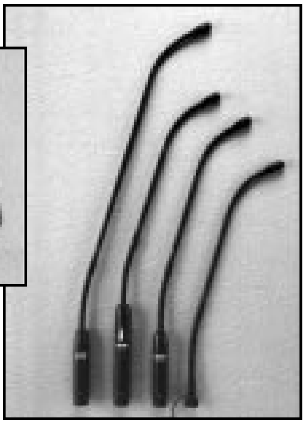


An unbalanced output signal is carried on a single conductor (plus a shield). An unbalanced input is sensitive to any signal on that conductor. Noise or hum which is picked up by the cable will be added to the original microphone signal and will be amplified along with it by the unbalanced input. For this reason, unbalanced microphones and cables are never recommended for long cable runs, or in areas of high electrical noise.

The two most common microphone output configurations (and mixer input configurations) are balanced low impedance and unbalanced high impedance. Since all high-quality and even most medium-quality microphones have balanced, low impedance outputs, this is the recommended configuration for the majority of meeting facility sound system applications.

5) Physical design:
How does the mechanical and operational design relate to the intended application?

Microphones for meeting facility sound applications include several typical designs: handheld; user-worn; free-standing mounted; and boundary or surface mounted. Each is characterized by a particular size, shape, or mounting method that lends itself to a specific manner of use. In addition, some microphones may be equipped with special features, such as on-off switches, that may be desirable for certain situations.



Handheld types are widely used for speech in many areas of meeting facility sound. Since they are usually handled, passed from person to person, or used while moving about, they must have a very effective internal shock mount to prevent pickup of handling noise. In addition, they are often used very close to the mouth and should therefore be equipped with an effective "pop" filter or windscreen. Size, weight, and feel are important considerations for a handheld microphone.

User-worn microphones include clip-on types that may be attached directly to clothing, lavalier styles worn on a lanyard around the neck, or possibly head-worn types. These are used when a relatively unobtrusive microphone must be located close to the user and especially for applications in which the wearer moves about or needs hands-free operation. Size and appearance are critical in the selection of user-worn microphones.

Free-standing mounted microphones (mounted away from large surfaces) come in a variety of styles suited for different fixed settings. These range from full-size microphones, on heavy-duty stands, to miniature types on unobtrusive goosenecks or booms, to hanging microphones of any size. Mounted microphones are generally selected for permanent installation although many handheld types may be placed in mounts and removed as needed. Shock isolation is still essential if the stand is likely to be moved or is mounted on a vibrating stage or hollow lectern. Windscreens are necessary for close-up speech or if used outdoors. Again, appearance is often a primary factor in the selection of mounted microphones.

Boundary or surface-mounted microphones are also used in fixed positions but the surface to which they are attached is integral to the operation of the microphone. These microphones are most successfully mounted on existing surfaces, such as tables, floors, walls, or ceilings, to cover a certain area. They depend to some extent on the acoustic properties of the mounting surface (size, composition, orientation) for their frequency response and directionality. However, they offer a very low profile and can minimize certain acoustic problems which are due to reflected sound. Appearance and physical environment play an important part in the selection of boundary microphones.

It should be noted that almost any combination of the four other microphone characteristics can be found in any of the physical designs mentioned here. That is, most of these designs are available in a choice of operating principles, frequency responses, directional patterns, and electrical outputs.

Though not intrinsically related to the other four areas of microphone specification, the physical design is no less important in the selection process and, indeed, is often one of the first choices dictated by the application. In any case, the other microphone specifications should be just as carefully chosen to satisfy the basic acoustic and electrical requirements of the application. Ultimately, all five areas must be properly specified to yield the best selection.

West Virginia Senate Caucus Room

Microphones: Use

Once a microphone is selected for a given application, it must be used properly to get the best possible results. Again, there are two key areas: the interface of the microphone with the sound source, and the interface of the microphone with the sound system. The first area involves primarily acoustic considerations for optimum placement of one or more microphones. The second area involves electrical and mechanical considerations for optimum operation of microphones.

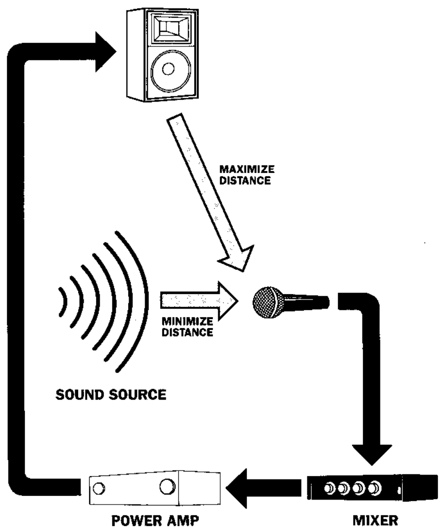
Microphone placement is a challenge that depends on the acoustic nature of the sound source and the acoustic characteristics of the microphone. Although this may appear to be a very subjective process, a description of some of the important acoustic interactions will lead to a few simple rules for successful microphone placement.

Recall that sounds can be categorized as desired or undesired and that the soundfield (total sound in a space) is made up of both direct sound and ambient sound. The level of direct sound decreases with distance (the inverse-square law) while ambient sound stays at a constant level. The critical distance is the distance from the sound source at which the level of direct sound has fallen to the level of the ambient sound. In practice, microphones must be placed much closer than the critical distance to get an acceptable ratio of direct-to-ambient sound.

This brings up the concept of distant pickup capability. The proportion of direct vs. ambient sound picked up by a microphone is a function not only of distance but of the directional pattern of the microphone as well. For a given ratio of direct-to-ambient sound, a unidirectional microphone may be used at a greater distance from the direct sound source than an omnidirectional type. This is called the distance factor, and ranges from about 1.7 for a cardioid, to 2.0 (twice the omnidirectional distance) for a hypercardioid.

For instance, if an omnidirectional microphone picked up an acceptable direct-to-ambient sound ratio at 2 feet from the sound source, then a cardioid would have the same ratio at about 3.4 feet, although the gain would have to be increased electronically to achieve the same output level. However, for a very weak source, or a very high ambient sound level, the acceptable omnidirectional location (again, less than the critical distance) could be as little as 3 inches away, for example. In this case, even the hypercardioid could only be used 6 inches away. This ill-named concept of microphone reach is very subjective and is dominated by the actual direct vs. ambient sound level at the microphone position rather than by the directionality of the microphone. Even an omnidirectional would have excellent reach, if no ambient sound were present!

In a sound reinforcement system, a second effect occurs with greater source-to-microphone distance: increased feedback potential. The farther the microphone is placed from the desired sound source, the more gain, or volume, is required to get adequate level. As the gain of a sound system is increased, amplified sound produced by the loudspeakers will be picked up by the microphone and re-enter the system.



Feedback path

At some setting, this re-entrant sound will be amplified to the same level as the original sound picked up by the microphone and the system will begin to ring or resonate. Higher settings (to compensate for greater microphone distance, for instance) will result in the sustained howl or squeal known as feedback. This condition depends not only on the system gain, but on the source-to-microphone distance, the microphone-to-loudspeaker distance, and the directionality of both the microphone and the loudspeaker.

This leads to the first and most important rule of microphone placement: Place the microphone as close as practical to the desired sound source. It has several corollaries: 1) place the microphone as far as possible from loudspeakers and other undesired sources; 2) use directional microphones to minimize ambient sound pickup; 3) aim directional microphones toward the desired sound and/or away from undesired sound; and 4) keep the system gain to a minimum.

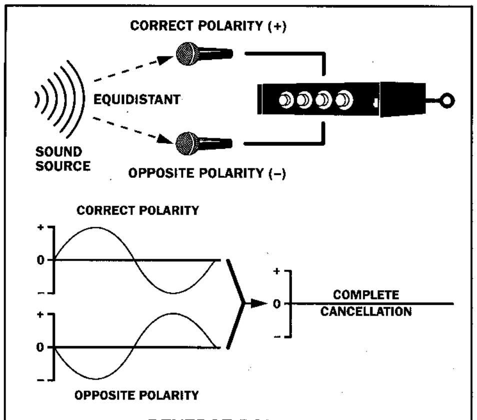
Ultimately, the position chosen should be consistent with the characteristics of both the sound source and the microphone: larger sources such as an audience may require greater distance, depending on the microphones' directionality; extremely loud sources may require greater distance to avoid overload of some sensitive condenser microphones; very short distances may cause proximity effect (low frequency boost) in some directional microphones; close-up speech use requires adequate "pop" filtering. In any case, following the above rules will give the best pickup of the desired sound, the minimum pickup of ambient noise, and least likelihood of feedback.

Another important acoustic interaction is called interference. Interference effects may occur whenever delayed versions of the same sound are mixed together, acoustically or electrically. With microphones, this may happen in several ways: 1) microphones of reverse polarity picking up the same sound, 2) multiple microphones picking up the same sound from different distances, 3) a single microphone picking up multiple reflections of the same sound, or 4) any combination of these. The effects are similar in each case, and include audible peaks and notches in frequency response, changes in directionality, and increased feedback problems.

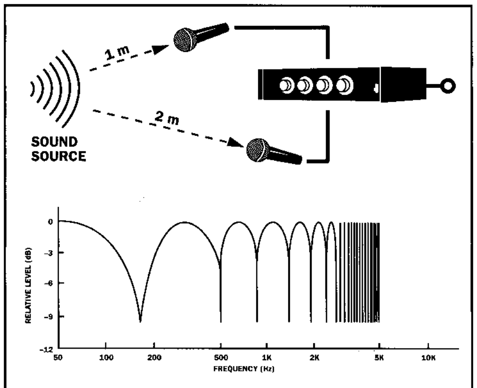
The first situation, reverse polarity, will result in severe loss of sound, especially low frequencies, when the microphones are placed next to each other and set to the same level. Signals from the microphones are then of equal strength but of opposite polarity. When these signals are combined in a mixer the cancellation is nearly total.

Although there is an international standard for microphone polarity (pin 2 +, pin 3 -), a reversal may be found in a mis-wired microphone or, more commonly, in a mis-wired microphone cable. It can be identified by checking each microphone and cable against a microphone and cable that are known to be correct. In any installation, all microphones and microphone cables must have the same polarity.

The second form of interference, multiple microphone pickup, can occur whenever more than one microphone is used. If the microphones are at unequal distances from the sound source, the sound picked up by the more distant microphone will be delayed relative to the nearer microphone. When these signals are combined in a mixer, peaks and notches occur at multiple frequencies which are related to the delay time, and hence, to the distance, between the microphones. This effect is called comb filtering because the resulting pattern of notches in the frequency response resembles the teeth of a comb. As the delay time increases, comb filtering begins at lower frequencies. It is especially noticeable at mid- and high-frequencies, and creates a "hollow", distant sound.

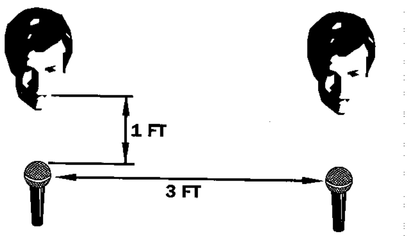


REVERSE POLARITY



MULTI-MICROPHONE PICKUP

The solution to this problem is to use the three-to-one rule: for multiple microphones, the microphone-to-microphone distance should be at least three times the source-to-microphone distance. For example, when using individual microphones for a city council, if a talker's microphone is one foot away, then the next nearest microphone should be at least three feet away from the first. This insures that direct sound from the talker will not be strong enough to cause noticeable interference when picked up by the more distant microphones. As the source-to-microphone distance increases, the distance to adjacent microphones must also be increased.



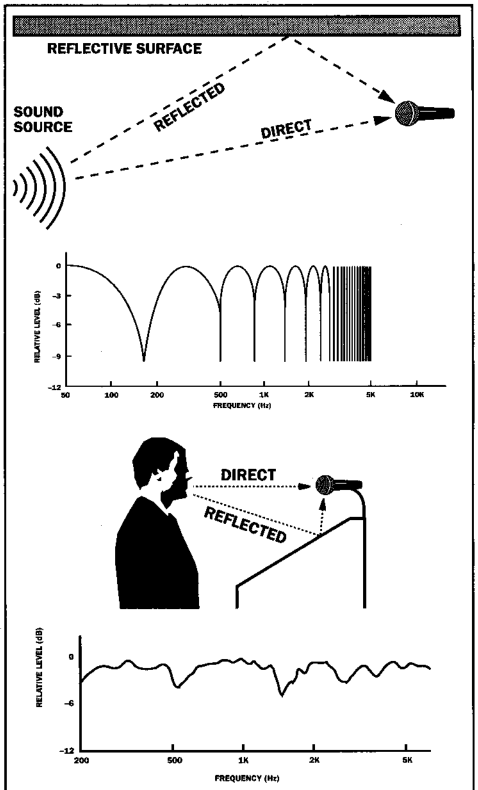
3 TO 1 RULE

An implication of the three-to-one rule is the following: avoid picking up the same sound source with more than one microphone. Microphones should be placed and aimed to minimize areas of overlapping coverage. This is important for a number of meeting facility sound applications. For audience pickup applications, each section or area should be covered by only one microphone. For lectern applications, only one microphone should be used. When a person wearing a lavalier microphone also speaks into a fixed microphone, one of the microphones should be turned down.

The third form of interference, reflection pickup, may occur whenever there are nearby sound-reflecting surfaces. This is often true in meeting facility settings: hardwood or stone floors, brick or glass walls, wood or plaster ceilings, and solid lecterns and desks. Recall that reflected sound is always delayed relative to the direct sound. When the delayed, reflected sound mixes with the direct sound at the microphone, comb filtering is again the result.

The first solution is to increase the direct sound level, by placing the microphone as close as practical to the sound source, so that the direct sound is much stronger than the reflected sound. Interference effects only become noticeable when the reflected sound is comparable in level to the direct sound. Close placement may not be possible in the case of area coverage, or moving sound sources.

The second solution is to decrease the reflected sound level. The microphone may be moved away from the reflective surface, or reoriented for minimum pickup of sound from that direction. The acoustically reflective surface may possibly be moved away, reoriented, or treated with some sound-absorbent material. However, this is often not feasible, for economic or aesthetic reasons.



REFLECTION PICKUP

The third alternative is to minimize the delay. Since the delay is due to the difference in the paths of the direct and reflected sound, this can be accomplished by moving the microphone so close to the reflective surface, that the direct sound and the reflected sound have nearly the same path. This raises the frequency at which comb filtering begins. If the microphone can be brought very close to the surface (within one-quarter inch), any comb filtering will occur above the audible range.

Surface-mount or "boundary effect" microphones are designed for this purpose. They effectively reduce interference from the surface to which they are attached. If they are located at the junction of two or more surfaces, such as the corner of a room, they reduce interference from each adjacent surface. In addition, a boundary microphone exhibits increased sensitivity due to addition of the direct and reflected sound.

To minimize reflective interference, avoid using microphones near acoustically reflective surfaces. If this is not possible, consider using a surface-mount microphone on the primary reflecting surface.

In addition to interference problems, the use of multiple microphones creates other potential difficulties. One of these is due to the fact that as the number of active microphones in a sound system increases, the overall potential for feedback also increases. And of course, each active microphone is adding more ambient noise pickup to the system.

This leads to a final general rule for microphone use: Always use the minimum number of microphones. If additional microphones are not needed, they may actually degrade the sound system. If the application can be satisfied with one microphone, use one microphone!

The second key area of microphone use is the interface of the microphone with the sound system. As mentioned at the beginning of this section, this involves primarily electrical considerations. We will develop a few simple rules for proper interface based on the electrical characteristics of the microphone output and the sound system input, and on the requirements for electrical and mechanical reliability.

In the discussion of operating principle it was mentioned that all condenser microphones require power for their operation. This is provided by an internal battery in some models, or by phantom power in others. If a condenser is selected, care must be taken to assure that the appropriate power source (battery or phantom) is available. A battery-powered condenser is fine for applications such as portable recording but phantom power should be always employed for any permanent installation of condenser microphones.

Phantom power is provided through the microphone cable itself. It is a DC (direct current) voltage that may range from 11 to 52 volts, depending on the microphone requirement and the phantom power source rating. This voltage is applied equally to the two conductors of a balanced microphone cable, that is, pin 2 and pin 3 of an XLR-type connector. The voltage source may be either in the mixer itself or in a separate phantom power supply connected in line with the microphone cable. Most recent mixers have phantom power built-in, and the actual voltage will be stated on the mixer or in the operating manual.

The voltage requirement for a phantom-powered condenser microphone will also generally be stated on the microphone or in the manufacturer's literature. Some types, particularly those that are externally charged, may require a full 48 volt supply. Electret types, which have a permanent charge, will typically operate over the entire range from 12 to 48 volts. Unless specifically stated otherwise by the manufacturer, these microphones will deliver their full performance at any voltage in this range, and further, they will not be damaged by a full 48 volt supply. Supplying less than the recommended voltage to either type may result in lower dynamic range, higher distortion, or increased noise but this also will not damage the microphone.

Dynamic microphones, of course, do not require phantom power. However, many mixers have only a single switch which supplies phantom power to all microphone inputs, which may include some used by dynamic microphones. The presence of phantom power has no effect on a balanced, low impedance dynamic microphone. It is not possible to damage or impair the performance of a balanced microphone which is correctly connected to any standard phantom supply.

If a balanced microphone is mis-wired or if an unbalanced, high-impedance microphone is used, there may be a loud "pop" or other noise produced when the microphone is plugged in or switched on. In addition, the sound of the microphone may be distorted or reduced in level. Even in these cases, the microphone will still not be damaged and will work normally when the wiring is corrected or the phantom power is turned off. If an unbalanced microphone must be used with a phantom-powered input, a transformer should be used to isolate it from the input. By the same token, it is also not possible to damage any standard phantom power source by improper microphone connection.

Good phantom power practices are: 1) check that phantom voltage is sufficient for the selected condenser microphone(s); 2) turn system levels down when connecting or disconnecting phantom-powered microphones, when turning phantom power on or off, or when turning certain phantom-powered microphones on or off; 3) check that microphones and cables are properly wired. Following these practices will make the use of condenser microphone almost as simple as dynamics.

Previously, it was suggested that, for the expected sound level, microphone sensitivity should be high enough to give a sufficient signal to the mixer input, but not so high as to cause input overload. In practice, however, most mixers are capable of handling a very wide range of microphone signal levels and sensitivity values are not critical. Occasionally, for extremely high sound levels, an attenuator may be necessary to lower the output of the microphone. These are built into some microphones and most mixers. Otherwise, accessory attenuators are available that may be inserted in line with the microphone cable.

It has already been mentioned that balanced, low impedance microphones are recommended for the majority of meeting facility sound applications. This will allow the use of long microphone cables, and result in the least pickup of electrical noise. In any case, the microphone impedance should be similar to the rated impedance of the microphone input of the mixer or other equipment. It is not necessary to match impedances precisely. It is only necessary that the actual mixer input impedance be greater than the microphone output impedance. In fact, the actual impedance of a typical mixer input is normally five to ten times higher than the actual output impedance of the microphone.

When it is necessary to match a balanced, low impedance microphone to an unbalanced, high impedance input, or vice versa, transformers with the appropriate input and output connectors are readily available. Transformers provide an impedance matching function, a level matching function, and can also change the configuration from balanced to unbalanced as needed. Ideally, transformers should be connected so that the bulk of the cable run is balanced, low impedance, for maximum length and minimum noise pickup. This would normally place the transformer at the connector of the unbalanced, high impedance device.

Professional (and most semi-professional) equipment has balanced, low impedance microphone inputs using 3 pin XLR-type connectors. Less sophisticated consumer electronic products and many tape recorders typically have unbalanced, high impedance microphone inputs using ¼" phone jacks or 3.5mm mini-phone jacks. A few mixers offer both types of connectors for each input channel. Simple adapters may be used to mate different types of connectors if no configuration change (high/low impedance or balanced/unbalanced signal) is necessary. Always use only high quality connectors and adapters.

Optimum microphone performance depends on the associated connectors and cables. In addition to quality connectors of the types described above, it is equally important to use high quality cables. Beyond the basic specification of balanced (two conductors plus shield) or unbalanced (one conductor plus shield), there are several other factors that affect the performance of cables.

The outer jacket protects the shield and conductors from physical damage and may be rubber or plastic. It should be flexible, durable, and abrasion resistant. Depending on the location it may need to be chemical or fire resistant. Different color jackets are available and can be used to identify certain microphone channels or cables. . . . . . . The shield protects the conductors from electrical noise. It may be braided or spiral-wrapped wire or metal foil. It should provide good electrical coverage and be flexible enough for the intended use: braid or spiral for movable use; foil only for fixed use such as in conduit.

The conductors carry the actual audio signal and are stranded or solid wire. They should be of sufficient size (gauge) to carry the signal, and provide adequate strength and flexibility. Use stranded conductors for most applications. Use solid conductors only for stationary connections.

A large percentage of “microphone” problems are actually due to defective or improper microphone cables. Microphone cables should be handled and maintained carefully for long life: 1) position them away from electrical sources to prevent hum or other noise pickup; 2) allow them to lie flat when in use to avoid snagging; 3) use additional cable(s) if necessary to avoid stress; 4) do not tie knots in cables; 5) coil loosely and store when not in use; and 6) periodically check cables visually and with a cable tester.

Individual, pre-assembled microphone cables are readily found in a wide variety of styles and quality. In addition, multiple cable assemblies, called snakes, are available for carrying many microphone signals from one location to another, such as from the courtroom to the equipment rack. The use of only high quality cables and their proper maintenance are absolute necessities in any successful meeting facility sound application.

Finally, the use of microphones in particular applications is often facilitated by microphone accessories. These are mechanical and electrical hardware items that are often used in mounting and connecting microphones.

Mechanical accessories include various kinds of acoustic devices such as windscreens and directionality modifiers. Windscreens, usually made of foam or cloth, should be used whenever microphones are used outdoors or subjected to other air currents or rapid motion. Pop filters are employed when the microphone is used close to the mouth, such as on lecterns or for handheld speech. These minimize noise caused by plosive consonants such as “p”, “b”, “t”, or “d”. Although such filters are usually supplied with microphones designed for these applications, additional protection may be needed in some cases or for microphones not already equipped. Use only high quality screens and filters to avoid degrading the sound of the microphone.

There are directional or polar modifiers available for certain microphones that can change the pickup pattern from cardioid to supercardioid, for example, or from omnidirectional to semi-directional in the case of some boundary microphones. Consult the manufacturer for proper use of these accessories.

Mounting accessories are of great importance in many meeting facility sound applications. Stands, booms, and goosenecks should be sturdy enough to support the microphone in the intended location and to accommodate the desired range of motion. Hanging hardware, to allow microphones to be suspended above an audience for example, must often include provision for preventing motion of the microphone due to air currents or temperature effects. Clips, or stand adapters, may be either permanent or designed for quick-release. Shock mounts are used to isolate the microphone from vibrations transmitted from the mounting surface, such as on a lectern.

Electrical accessories such as transformers and phantom power supplies have already been described. In addition, there are a variety of signal processors which may be used directly in line with a microphone. These can range from simple low or high frequency filters, to complete preamp/equalizer/limiter units, though most of these functions are normally provided by the mixer and subsequent elements of the audio chain.

Creative use of these accessories can allow microphones to be mounted with acceptable aesthetic appearance. Again, quality and reliability should be primary concerns.

Wireless Microphone Systems

A wireless microphone is actually a system consisting of 1) a microphone 2) a radio transmitter and 3) a radio receiver. The function of the microphone is unchanged and the function of the transmitter and receiver combination is to replace the microphone cable with a radio link. Although this objective is simple, its accomplishment is not. However, with some knowledge of the components and characteristics of wireless microphone systems, and a clear idea of the intended application, the selection and use of wireless microphones can be made relatively straightforward.

Wireless microphone system components

1) The Microphone:
   H*ow does sound enter the wireless system?*

The selection process for the microphone part of a wireless system is exactly the same as for wired microphones. The microphone must be matched to the desired sound source and to the sound system. In this case, the sound system consists not only of the devices that make up the rest of the audio chain but the input to the radio transmitter as well. Acoustically, wireless and wired microphones behave identically. Proper microphone choice and placement is still necessary to get the best sound and to avoid problems such as feedback.

Available microphone choices for wireless include dynamic or condenser types, with flat or shaped frequency response, omnidirectional or unidirectional polar patterns, and a variety of physical designs: lavalier, handheld, headworn, etc. Almost any type of microphone may be used as part of a wireless system, the notable exception being phantom-power-only condensers. The choice depends on the specific application.

2) The Transmitter:
   How does the microphone signal become a radio signal?

The transmitter uses the audio signal from the microphone to vary the frequency of a radio signal which is broadcast to the receiver. The principle is called Frequency Modulation (FM) and is identical to that used by commercial FM radio stations. Electrically, the transmitter input must be compatible with the microphone output both in level and impedance. The transmitter input may also supply bias power for some condenser microphone elements. (Note that bias power is not the same as phantom power.) The transmitter itself is always battery-powered.

Physically, the transmitter takes one of two forms. The first is a small box, called a body-pack or belt-pack, that can be clipped to a belt, placed in a pocket, or otherwise attached to the user. The microphone connects to it by means of a small cable. Some models have a detachable cable which allows the transmitter to be used with a variety of inputs. This form is most often used with lavalier microphones but can also be connected to head-worn microphones or hand-held types with appropriate cables. All transmitters have a power on-off switch and many have a mute switch to silence the microphone without turning off the radio signal itself.

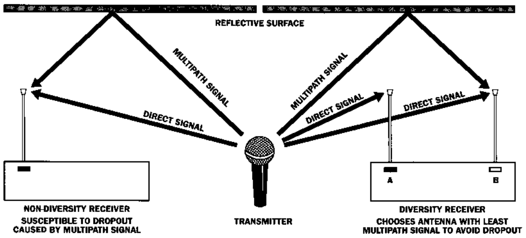
The second form is a transmitter that is built into the cylindrical body of the microphone itself. This is used almost exclusively for handheld speech microphones and results in a package only slightly larger than a conventional wired microphone.

3) The Receiver:
How is the radio signal turned back into an audio signal?

The receiver picks up the radio signal broadcast by the transmitter and extracts or demodulates the audio signal from it. Again the principle is the same as that of an ordinary FM radio. The output of the receiver is electrically identical to a microphone output and can be connected to any typical microphone input in a sound system. Some receivers have additional amplified outputs for headphones or auxiliary connections to sound systems. Although most receivers operate on ordinary 120V or 230V AC power, battery-powered types are available for portable use.

Wireless receivers are also designed in two different configurations. The first is called non-diversity and consists of a single antenna and a single radio circuit. An ordinary FM radio is an example of a simple, non-diversity receiver.

Non-diversity receivers work well in many applications but are subject to a phenomenon known as dropout: a temporary interruption of the radio signal. The audible effect may range from a slight "swishing" noise to a complete loss of sound. Since radio signals become weaker over greater distances, a dropout can occur when the transmitter is very far from the receiver antenna. A dropout can also occur at shorter distances when the radio signal from the transmitter is blocked by obstacles such as human bodies, walls, or equipment. Finally, dropouts may be experienced even at very short distances by a mechanism called multipath interference. Part of the signal from the transmitter (which radiates in all directions) travels directly to the receiver, but some of the signal is reflected to the receiver by metal objects or other structures. When the paths of the direct signal and of the reflected signal(s) are sufficiently different, they will interfere with each other when they combine at the receiver antenna. If the interference is great enough, partial or complete cancellation of the signal occurs, resulting in a dropout. It is similar to an extremely severe ghost in television reception, and the cure is the same: move the receiver antenna relative to the transmitter. This is not usually practical since it is the receiver antenna that is in a fixed location, while the wireless microphone location is constantly changing.



This introduces the concept behind the second wireless receiver configuration, called a diversity system. A diversity receiver utilizes two separate antennas and (usually) two separate radio circuits. When the two antennas are separated by even a short distance, the chance of a simultaneous interruption at both antenna positions is extremely low. The key to the system is additional intelligent diversity circuitry which continuously monitors the received signals and takes action according to the type of diversity circuitry employed.

The simplest effective diversity technique, known as "phase" diversity, has two antennas but only one radio circuit. The diversity circuitry adjusts the relative polarity of the antennas before combining them for optimum reception. This approach is less expensive, due to the single radio section, and works well when both antennas are getting a usable signal. But, it may not give the best results in the case of severe multipath interference.

Most "true" diversity receivers are of the switching type. These utilize two antennas and two radio sections. The diversity circuitry selects the better of the two received signals by means of an electronic switch. If the switching is done quickly and quietly enough, the result is nearly dropout-free performance, with minimal audible side effects.

The third diversity design is known as the combining type. This method takes advantage of the fact that both of the received signals are usable much of the time. In this case, using the signals from both antennas yields better reception than using only one signal (as in the switching type). The combining diversity circuitry adds the signals proportional to their relative strength. When both are strong, the contribution from each signal is equal. If one signal becomes weaker, its contribution is similarly reduced. Finally, if a complete dropout occurs for one signal, the receiver uses only the good signal. Since the combining technique acts as a continuous balance control rather than as a switch, it further reduces any audible effects of diversity action.

Historically, diversity receivers have always been used for critical applications even though their cost was somewhat higher. Today, the cost of wireless systems in general and diversity systems in particular has decreased to the point that diversity receivers are used in the majority of meeting facility applications.

An additional refinement in nearly all recent wireless systems is some form of noise reduction, or companding, in order to decrease the inherent noise and increase the limited dynamic range of radio transmission. The word companding refers to the two steps of the process: the signal is modified (compressed) in the transmitter before it is broadcast and then "un"-modified (expanded) in the receiver in a complementary fashion. Although the principle of companding is similar in all wireless systems, significant differences between manufactured models make it undesirable to mix transmitters of one brand with receivers of another brand.

Other aspects of wireless microphone systems that must be considered in selection and use are operating frequencies, antennas, and radio interference. All three are especially important when planning the use of multiple wireless systems in the same location.

Every wireless microphone system transmits and receives on a particular radio frequency, called the operating frequency. These frequencies may be grouped into four bands: low-band VHF (49-72 MHz), high-band VHF (169-216 MHz), low-band UHF (450-614 MHz) and high-band UHF (806-952 MHz). VHF stands for "Very High Frequency"; UHF stands for "Ultra High Frequency"; and MHz stands for "MegaHertz" or millions of cycles-per-second. Use of these bands is regulated by the Federal Communication Commission (FCC). Certain frequencies within each band have been designated for use by wireless microphones as well as by other devices. It should be noted that while manufacturers must be licensed by the FCC to sell wireless equipment, it is the responsibility of the purchaser to observe FCC regulations regarding their actual use.

Low-band VHF, particularly 49 MHz, is shared not only by consumer grade wireless microphones but by cordless telephones, walkie-talkies, and radio controlled toys. For this reason, it is almost never recommended for serious applications, even though systems in this range are very inexpensive.

The high-band VHF range is the most widely used for professional applications, and quality systems are available at a variety of prices. Although this range includes U.S.

television channels 7-13, there are ample frequencies for use in any part of the country.

The two UHF bands, while free of many interfering signals found in the VHF bands, have traditionally been used only by very costly systems, due to more complex design and circuitry. However, as the price of UHF systems declines and the VHF band becomes more crowded, the UHF band has become increasingly attractive.

Selection of operating frequency for a single wireless system only involves choosing a frequency that is unused by local television stations. Although there are tunable or frequency agile systems available, most equipment operates at a fixed frequency controlled by a quartz crystal which is installed and adjusted by the manufacturer. Most manufacturers can recommend a frequency that will generally work in a given geographic area. The difficulty arises when multiple systems must work together in the same location.

Due to the nature of radio reception, it is not possible for a single receiver to pick up multiple transmitters on the same frequency. Therefore, each transmitter must be on a separate frequency and have a corresponding receiver on that frequency. An additional complication is that simultaneously operating systems, even though they may be on different frequencies, may interfere with each other if those frequencies are not carefully chosen. The rules for frequency selection are complex enough that computer programs are used to calculate compatible systems. Again, the wireless microphone manufacturer should be consulted for any multiple system installation.

The selection of a wireless microphone system includes several steps, some of which are similar to wired microphone selection. It should be remembered that while wireless microphones cannot ultimately be as consistent and reliable as wired microphones, the performance of present systems can be very good, allowing excellent results to be obtained. Following these steps will help select the best system(s) for your application.

First, define the application. In a meeting facility system, this may be a wireless lavalier microphone for a presenter and a wireless handheld microphone for audience questions.

Second, choose the microphone type. The application will usually determine which microphone type is required: a lavalier or clip-on type attached to clothing, or a head-worn type, both for hands-free use; a handheld type for when the microphone must be passed around to different users. Most handheld and headworn types are unidirectional, while lavaliers are usually omnidirectional. Unidirectional lavaliers are available, for use when feedback or high ambient noise is a problem.

Third, choose the transmitter type. Again, the application will specify the choice. All but the handheld type will use some kind of body-pack transmitter. Some body-pack transmitters, especially those with a multi-use input connector, use a separate antenna wire while others use the permanently attached microphone cable as the antenna. A mute or audio on-off switch is desirable to avoid turning off the transmitter power when the microphone is not needed. Handheld types may have external or internal antennas. Transmitter batteries may be one of several types and their relative availabilty should be considered. Also, power consumption of transmitters varies so be aware of expected battery life.

Fourth, choose the receiver type. The basic choice here is diversity vs. non-diversity. For reasons mentioned in the receiver section above, diversity receivers are recommended for all but the most budget-conscious applications. Non-diversity types will work well in most situations, but the extra insurance and extra features of the diversity receiver are often worth the slightly higher cost. Other features of the receiver, such as headphone outputs, balanced outputs, different indicators, and potential for battery power may be desirable.

Fifth, determine the number of systems to be used. This should take into account future additions to the system. Choosing a system that can only accommodate a few frequencies may be a future limitation. It must also take into account existing wireless systems with which the new equipment must work.

Sixth, consult the manufacturer or a knowlegeable professional about frequency selection to integrate the planned number of systems. This must be done for any multiple system installation and should be done for even single systems to avoid potential interference problems.

Once the wireless system(s) choice is made, good installation and proper use are necessary for satisfactory performance. Antenna selection and placement are very important aspects of installation. There are a few general rules about antennas to keep in mind.

First, maintain line-of-sight between the transmitter and receiver antennas if possible. Avoid human bodies, metal, or other dense materials between the two.

Second, keep the distance from transmitter to receiver as short as possible. It is much better to have the receiver near the transmitter and run the audio signal from the receiver through a long cable than to transmit over long distances or use long antenna cables. The maximum legal signal strength of VHF systems is only fifty one-thousandths of a watt (0.050 W)! This is a very tiny, tiny signal.

Third, use the proper receiver antenna. A 1/4-wave antenna (about 17 inches long for high-band VHF) can be used if it is mounted directly to the receiver. If the antenna is to be located at a distance from the receiver, which will be neccessary if the receiver is mounted inside a metal enclosure or at a great distance from the transmitter, a 1/2-wave or other high gain antenna should be used.

Fourth, mount antennas vertically and away from large metal objects. This applies to receiver and to transmitter antennas. Do not coil or fold up trailing wire antennas, or microphone cable antennas, on body-pack transmitters.

Fifth, use the proper antenna cable for remotely locating receiver antennas. The correct impedance is usually 50 ohms. Employ the minimum length necessary. Use low-loss cable for longer cable runs.

Sixth, mount multiple antennas at least 1/4 wavelength (about 17 inches) apart. Use an amplified antenna distribution system (sometimes called an "active" antenna splitter) to minimize the number of antennas and to reduce interference problems with multiple receivers. This allows one antenna or one antenna pair (for a diversity system) to be used with multiple receivers.

The last aspect of the use of wireless microphone systems and the least predictable is radio interference. Potential interference from other wireless systems operating on the same or nearby frequencies has been discussed, but what about other possible sources of interference? High-band VHF systems are generally not subject to interference from radio stations, amateur radio, pagers or cellular telephones. Television stations 7-13 are a possibility but are very predictable. Always avoid wireless microphone frequencies within the bands of locally active TV channels. So-called travelling frequencies (169 to 172 mHz) are available that are just below the VHF television band. Keep in mind that these travelling frequencies may be also used for business two way radio or government use. If so, there will likely be interference problems.

Unpredictable interference sources include the following: 1) any type of digital device such as computers, digital signal processors, DAT or CD players; 2) neon or fluorescent light fixtures; 3) large motors and generators; 4) any electrical device that uses high voltage or high current; or 5) any device that is marked with an FCC type rating sticker.

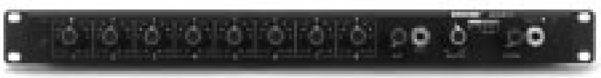
Good practice with any wireless system is to check out the system ahead of meeting time with all other systems and devices activated. This will reveal potential problems that were not apparent in a wireless system only test.

Many wireless receivers are equipped with an adjustable squelch control. This sets the basic sensitivity of the receiver to avoid picking up interfering signals, or background radio noise, when the transmitter is turned off or if a dropout occurs. It should be adjusted according to the manufacturers' instructions.

Ideally, transmitters should be turned on first, then receivers. Once the system is on, use the mute or mic switch to turn off the transmitter audio if necessary. Do not turn off the transmitter until after the event is over and/or the receiver is turned off. This will avoid an "open" receiver, which can pick up other radio signals that may be present.

Finally, always use fresh batteries of the correct type in the transmitter! Most manufacturers recommend only alkaline or lithium type batteries for adequate operation. Use rechargeable batteries with caution as their actual voltage is usually less than stated, and they may not operate satisfactorily in a wireless transmitter. In addition, the actual operating time of a rechargeable battery is normally much less than an alkaline type.

Automatic Mixers



Automatic Microphone Mixer

The function of an automatic mixer is twofold: 1) to automatically activate microphones as needed and 2) to automatically adjust the system gain in a corresponding manner. With some automatic mixers, ordinary microphones are used and the actual control is provided by the mixer. In others, special microphones are integrated with the mixer to provide enhanced control.

Scott Air Force Base

The reasons for using an automatic mixer relate to the behavior of multiple microphone systems. Each time the number of open or active microphones increases, the system gain also increases. This results is a greater potential for feedback as more microphones are added, just as if the master volume control were being turned up. Also, when multiple open microphones pick up the same talker, a degradation of audio quality occurs, called comb filtering. Since sound travels at a finite speed, the talker's voice arrives at the microphones at different times. When electronically combined in a mixer, these "out-of-step" microphone signals produce a combined frequency response very different from the frequency response of a single microphone. The aural result of comb filtering is an audio signal that sounds hollow, diffuse, and thin. In addition, unwanted background noise increases with the number of open microphones. Here, the effect is a loss of intelligibility as the background noise level rises closer to the level of the desired sound.

The solution is to activate microphones only when they are addressed and to keep them attenuated (turned down) when not being addressed. In addition, when more than one microphone is addressed at a time, the system gain must be reduced appropriately to prevent feedback and insure minimum noise pickup.

There are several techniques used to accomplish channel activation or gating in an automatic mixer. In most mixers, a microphone is gated on when the sound it picks up is louder than some threshold or reference level. When the sound level falls below the threshold, the microphone is gated off. This threshold may be fixed, adjustable, or even automatically adjustable.

Certain other automatic mixers, with integrated microphones, can actually sense the location of the sound source relative to the ambient noise and activate microphones only when the sound comes from the desired direction. These directional gating systems do not require any threshold adjustments.

There is another circuit within every automatic mixer that continuously senses the number of open microphones (NOM) and adjusts the gain of the mixer accordingly. With a properly functioning automatic system, if each individual microphone is adjusted to a level below the feedback point, then any combination of microphones will also be below the feedback point.

Summary of automatic mixer benefits

- The primary function of an automatic mixer is to keep unused microphones turned off and to instantaneously activate microphones when needed
- Using an automatic mixer will:
  - —Improve gain before feedback
  - —Reduce audio degradation caused by superfluous open microphones
  - —Control the build-up of background noise
- Keeping the number of open microphones to a minimum always improves overall audio and quality
- The additional control circuitry on automatic mixers provide a variety of additional functions like:
  - —Audio privacy switches
  - —Chairperson control of all microphones
  - —Illuminated indicators of microphone status
  - —Automatic video camera selection based on microphones activation

Many automatic mixers have additional control circuitry, often in the form of logic connections. These are electrical terminals that can be used for a variety of functions, including: microphone status indicators, mute switches, loudspeaker attentuation, and the selection of priority channels. Some automatic mixers have a selectable off-attenuation control: instead of gating the microphone completely off, it can be turned down by some finite amount which makes the gating effect less noticeable in certain applications. Another control included on some units is an adjustable hold time: when the desired sound stops, the channel is held on for a short time to avoid gating off the microphone between words or short pauses. Finally, most automatic mixers are able to be expanded by adding individual channels and/or by linking multiple mixers together to control large numbers of microphones simultaneously.

An automatic mixer must be considered whenever multiple microphones (four or more) are being used, particularly if the sound system is intended to run hands-free, that is, without a live operator. Microphones should be selected and placed according to the guidelines previously discussed. Automatic mixer systems with integrated microphones require a choice from the microphone models available for those systems. It is recommended that the manufacturer or a qualified sound contractor be consulted on the details of a particular automatic mixer.

Typical Applications

In order to select a microphone for a specific application, and to apply it properly, it is first necessary to know the important characteristics of the sound source(s) and of the sound system. Once these are defined, a look at the five areas of microphone specifications will lead to an appropriate match. Finally, proper use of the microphone, by correct placement and operation, will insure best performance. This section presents recommendations for some of the most common meeting facility sound applications.

Lectern

The desired sound source, for a lectern microphone, is a speaking voice. Undesired sound sources that may be present are nearby loudspeakers (possibly overhead) and ambient sound (possibly ventilation, traffic noise, and reverberation). The sound system in this and the following examples is assumed to be high quality with balanced low-impedance microphone inputs.

The basic performance requirements for a lectern microphone can be met by either dynamic or condenser types, so the choice of operating principle is often determined by other factors, such as appearance. In particular, the desire for an unobtrusive microphone is best satisfied by a condenser microphone, which can maintain high performance even in very small sizes. If phantom power is available, a condenser is an excellent choice. If not, dynamic types, though somewhat larger, are available with similar characteristics.

For the microphone to match the desired sound source (the talker's voice) it must first have a frequency response which covers the speech range, (approximately 100Hz to 10kHz). Within that range the response can be flat, if the sound system and the room acoustics are very good, but often a shaped response will improve intelligibility. Above 10kHz and below 100Hz, the response should roll off smoothly, to avoid pickup of noise and other sounds outside of the speech range, and to minimize proximity effect.

The choice of microphone directionality that will maximize pickup of the voice and minimize undesired sounds, is unidirectional. This type will also reduce the potential of feedback since it can be aimed toward the talker and away from loudspeakers. Depending on how much the person speaking moves about, or on how close the microphone can be placed, a particular type may be chosen: a cardioid for moderately broad, close-up coverage; a supercardioid or a hypercardioid for progressively narrower or more distant coverage.

The electrical characteristics of the microphone are primarily determined by the sound system: in this case, a balanced low-impedance type would match the inputs on the mixer. Of course, this would be the desired choice in almost all systems due to the inherent benefits of lower noise and longer cable capability. The sensitivity of the microphone should be in the medium-to-high range since the sound source (speaking voice) is not excessively loud and is picked up from a slight distance. Again, this is most easily accomplished by a condenser type.

The choice of physical design for a lectern microphone must blend performance with actual use. The most effective approach is a gooseneck-mounted type, which places the microphone close to the sound source and away from both the reflective surface of the lectern and noise from the handling of materials on it. Another approach is the use of a boundary microphone on the lectern surface, but this method is limited by lectern design and by the potential for noise pickup. As mentioned above, the desired physical design may also suggest the operating principle. The most effective small gooseneck or boundary styles are condensers.

The ideal placement of a lectern microphone is 6 to 12 inches away from the mouth, and aimed toward the mouth. This will give good pickup of the voice and minimum pickup of other sources. Also, locating the microphone a few inches off-center will reduce breath noise that might occur directly in front of the mouth. It is not recommended that two microphones be used on a lectern as comb filtering interference is likely to occur.

Proper operation of the microphone requires correct connection to the sound system with quality cables and connectors, and correct phantom power if a condenser is used. Use a shock mount to control mechanical noise from the lectern itself. Some microphones are equipped with low-cut or low-end roll-off filters, which may further reduce low frequency mechanical or acoustic noise. Goosenecks should be quiet when flexed. It is strongly recommended that a pop filter be placed on the microphone to control explosive breath sounds, especially when using miniature condenser types.

Good technique for lectern microphone use includes:

- Do adjust the microphone position for proper placement.
- Do maintain a fairly constant distance of 6 to 12 inches.
- Don't blow on microphone, or touch microphone or mount when in use.
- Don't make excess noise with materials on lectern.
- Do speak in a clear and well-modulated voice.

Meeting Table

The desired sound source at a meeting table, is a speaking voice. Undesired sounds may include direct sound, such as an audience or loudspeakers, and ambient noise sources such as building noise or the meeting participants.

A boundary microphone is the physical design best suited to this application. This will minimize interference effects due to reflections from the table surface and will also result in increased microphone sensitivity. A condenser type is the most effective for this configuration, due to its high performance and small size.

The frequency response should be slightly shaped for the vocal range and will usually benefit from a slight presence rise. A unidirectional (typically, a cardioid) pattern will give the broadest coverage with good rejection of feedback and noise. Finally, the microphone should have a balanced low-impedance output, and moderate-to-high sensitivity.

Placement of the microphone should be flat on the table, at a distance of two to three feet from, and aimed towards the normal position of the talker. If possible, it should be located or aimed away from other objects and from any local noise such as page turning. If there is more than one distinct position to be covered, position additional microphones according to the 3-to-1 rule.

The microphone should be connected and powered (if necessary) in the proper fashion. If the table itself is a source of noise or vibration, isolate the microphone from it with a thin foam pad. A low-frequency filter may be a desirable or even necessary feature. A pop filter is not normally required. Make certain the microphones are never covered with papers.

Good technique for meeting table microphone use includes:

- Do observe proper microphone placement.
- Do speak within coverage area of the microphone.
- Don't make excess noise with materials on table.
- Do project the voice, due to greater microphone distance.

Hand-held Speech Microphone
The desired sound source, for a hand-held microphone, is a speaking voice. Undesired sounds may include loudspeakers, other talkers, ventilation noise, and other various ambient sounds.

Suitable microphone performance for this application can be provided by dynamics or condensers. Due to frequent handling and the potential for rough treatment, dynamic microphones are most often used, though durable condensers are also available. The preferred frequency response is shaped with a presence rise for intelligibility and low roll-off for control of proximity effect and handling noise. These microphones are typically unidirectional. A cardioid pattern is most common, while supercardioid and hypercardioid types may be used in difficult noise or feedback situations.

Balanced low-impedance output configuration is standard while sensitivity may be moderate-to-low due to the higher levels from close-up vocal sources. Finally, the physical design is optimized for comfortable hand-held use, and generally includes an integral windscreen/pop filter and an internal shock mount. An on-off switch may be desirable in some situations.

Placement of hand-held microphones at a distance of four to twelve inches from the mouth, aimed towards it, will give good pickup of the voice, relative to other sources. In addition, locating the microphone slightly off-center, but angled inward, will reduce breath noise.

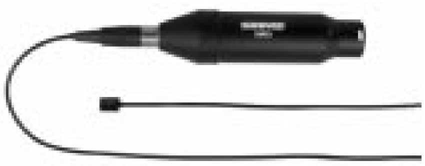
With high levels of unwanted ambient noise, it may be necessary to hold the microphone closer. If the distance is very short, especially less than four inches, proximity effect will greatly increase the low frequency response. Though this may be desirable for many voices, a low frequency roll-off may be needed to avoid a "boomy" or "muddy" sound. Additional pop filtering may also be required for very close use.

Use of rugged, flexible cables with reliable connectors is an absolute necessity with hand-held microphones. A stand or holder should also be provided if it is desirable to use the microphone hands-free. Finally, the correct phantom power should be provided if a condenser microphone is used.

Good technique for hand-held microphone use includes:

- Do hold microphone at proper distance for balanced sound.
- Do aim microphone toward mouth and away from other sounds.
- Do use low frequency roll-off to control proximity effect.
- Do use pop filter to control breath noise.
- Don't create noise by excessive handling.
- Do control loudness with voice rather than moving microphone.

Lavalier
The desired sound source, for a lavalier microphone, is a speaking voice. Undesired sources include other talkers, clothing or "movement" noise, ambient sound, and loudspeakers.



A condenser lavalier microphone will give excellent performance in a very small package, though a dynamic may be used if phantom power is not available or if the size is not critical. Lavalier microphones have a specially shaped frequency response to compensate for off-axis placement (loss of high frequencies), and sometimes for chest resonance (boost of middle frequencies). The most common polar pattern is omnidirectional, though unidirectional types may be used to control excessive ambient noise or severe feedback problems. However, unidirectional types have inherently greater sensitivity to breath and handling noise.

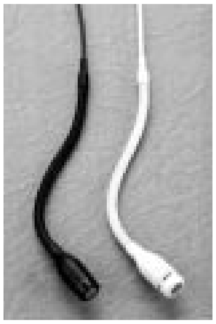
Balanced low-impedance output is preferred as usual. Sensitivity can be moderate, due to the relatively close placement of the microphone. The physical design is optimized for body-worn use. This may be done by means of a clip, a pin, or a neck cord. Small size is very desirable. For a condenser, the necessary electronics are often housed in a separate small pack, also capable of being worn or placed in a pocket. Some condensers incorporate the electronics directly into the microphone connector. Provision must also be made for attaching or routing the cable to minimize interference with movement. Wireless versions simplify this task.

Placement of lavalier microphones should be as close to the mouth as is practical, usually a few inches below the neckline on a lapel, a tie, or a lanyard, or at the neckline in the case of a woman's dress. Omnidirectional types may be oriented in any convenient way, but a unidirectional type must be aimed in the direction of the mouth.

Avoid placing the microphone underneath layers of clothing or in a location where clothing or other objects may touch or rub against it. This is especially critical with unidirectional types. Locate and attach the cable to minimize pull on the microphone and to allow walking without stepping or tripping on it. A wireless lavalier system eliminates this problem and provides complete freedom of movement. Again, use only high quality cables, and provide phantom power if required.

Good technique for use of lavalier microphones includes:

- Do observe proper placement and orientation.
- Do use pop filter if needed, especially with unidirectional.
- Don't breathe on or touch microphone or its cable.
- Don't turn head away from microphone.
- Do mute lavalier mic when using lectern or table microphone.
- Do speak in a clear and distinct voice.



Audience

The desired sound source is a group of talkers. Undesired sound sources may include loudspeakers and various ambient sounds.

The use of audience microphones is governed, to some extent, by the intended destination of the sound. In general, high level sound reinforcement of the audience in a meeting facility is not recommended. In fact, it is impossible in most cases, unless the audience itself is acoustically isolated from the sound system loudspeakers. Use of audience microphones to cover the same acoustic space as the sound system loudspeakers results in severe limitations on gain before feedback. The absolute best that can be done in this circumstance is very low level reinforcement in the immediate audience area, and medium level reinforcement to distant areas, such as balconies or foyers. Destinations such as isolated listening areas, recording equipment, or broadcast audiences, can receive higher levels because feedback is not a factor in these locations.

A condenser is the type of microphone most often used for audience applications. They are generally more capable of flat, wide-range frequency response. The most appropriate directional type is a unidirectional pattern, usually a cardioid. A supercardioid or a hypercardioid may be used for slightly greater ambient sound rejection. Balanced low-impedance output must be used exclusively and the sensitivity should be high because of the greater distance between the source and the microphone. This higher sensitivity is also easier to obtain with a condenser design.

The physical design of a microphone for audience pickup should lend itself to some form of overhead mounting, typically hanging. It may be suppported by its own cable or by some other mounting method. Finally, it may be a full size microphone, or a miniature type for unobtrusive placement.

A particular method that is sometimes suggested for overhead placement is a ceiling-mounted microphone, usually a boundary microphone. This position should be used with caution, for two reasons. First, it often places the microphone too far from the desired sound source, especially in the case of a high ceiling. Second, the ceiling, in buildings of modern construction, is often an extremely noisy location, due to air handling noise, lighting fixtures, and building vibration. Remember that a microphone does not reach out and capture sound. It only responds to the sound that has travelled to it. If the background noise is as loud or louder at the microphone than the sound from the talker below, there is no hope of picking up a usable sound from a ceiling-mounted microphone.

Placement of audience microphones falls into the category known as area coverage. Rather than one microphone per sound source, the object is to pick up multiple sound sources with one (or more) microphone(s). Obviously, this introduces the possiblity of interference effects unless certain basic principles, such as the "3-to-1 rule"are followed.

For one microphone, picking up a typical audience, the suggested placement is a few feet in front of, and a few feet above, the heads of the first row. It should be centered in front of the audience and aimed at the last row. In this configuration, a cardioid microphone can cover up to 20-30 talkers, arranged in a rectangular or wedge-shaped section.

For larger audiences, it may be necessary to use more than one microphone. Since the pickup angle of a microphone is a function of its directionality (approximately 130 degrees for a cardioid), broader coverage requires more distant placement. As audience size increases, it will eventually violate the cardinal rule: place the microphone as close as practical to the sound source.

In order to determine the placement of multiple microphones for audience pickup, remember the following rules: 1) the microphone-to-microphone distance should be at least three times the source-to-microphone distance (3-to-1 rule); 2) avoid picking up the same sound source with more than one microphone, and 3) use the minimum number of microphones.

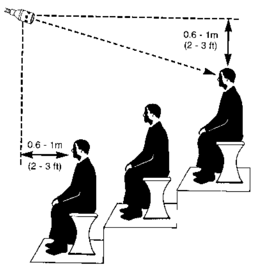
For multiple microphones, the objective is to divide the audience into sections that can each be covered by a single microphone. If the audience has any existing physical divisions (aisles or boxes), use these to define basic sections.

If the audience is a single large entity, and it becomes necessary to choose sections based solely on the coverage of the individual microphones, use the following spacing: one microphone for each lateral section of approximately 8 to 10 feet. If the audience is unusually deep (more than 6 or 8 rows), it may be divided into two vertical sections of several rows each, with aiming angles adjusted accordingly. In any case, it is better to use too few microphones than too many.

Once hanging microphones are positioned, and the cables have been allowed to stretch out, they should be secured to prevent turning or other movement by air currents or temperature changes. Fine thread or fishing line will accomplish this with minimum visual impact. Use only high quality cables and connectors, particularly if miniature types are specified.

Many older meeting facilities are very reverberant spaces, which provide natural, acoustic reinforcement for the audience, though sometimes at the expense of speech intelligibility. In spaces like this, it is often very difficult to install a successful sound system as the acoustics of the space work against the system. Most well-designed modern architecture has been engineered for a less reverberant space, both for greater speech intelligibility, and to accommodate modern forms of multimedia presentations. This results in a greater reliance on electronic reinforcement.

The use of audience microphones is normally exclusively for recording, broadcast, and other isolated destinations. It is almost never intended to be mixed into the sound system for local reinforcement. If it is desired to loudly reinforce an individual member of the audience, it can only be done successfully with an individual microphone placed amid the meeting participants: a stand-mounted type that the member can approach or a



Microphone positioning for audience pick-up.

hand-held type (wired or wireless) that can be passed to the member.

Good technique for use of audience microphones includes:

- Do place the microphones properly.
- Do use minimum the number of microphones.
- Do turn down unused microphones.
- Don't attempt to "over-amplify" the audience.
- Do speak in a strong and natural voice.

Non-meeting Applications

Today, the life of meeting facilities extends far beyond just meetings, to include classes, plays, and social events. Sound systems can play an important role in all of these situations. While it is not possible to detail microphone techniques for every application, a few examples will show how to use some of the ideas already presented.

Though most classrooms are not large enough to require the use of a sound system, it is sometimes necessary to record a class, or to hold a very large class in an auditorium. In these cases, it is suggested that the teacher wear a wireless lavalier microphone to allow freedom of movement and to maintain consistent sound quality. If it is desired to pick up the responses of students, it is possible to use area microphones in a recording application, but not with a sound system. A better technique is for questions to be presented at a fixed stand microphone, or to pass a wireless microphone to the student.

Microphone use for plays and other theatrical events involves both individual and area coverage. Professional productions usually employ wireless microphones for all the principal actors. This requires a complete sytem (microphone, transmitter, receiver) for each person, and the frequencies must be selected so that all systems will work together without interference. While it is possible to purchase or rent a large number of wireless systems, it is often more economical to combine just a few wireless systems with area microphones for the rest of the players. Use unidirectional boundary microphones for "downstage" (front) pickup, and use unidirectional hanging microphones for "upstage" (rear) pickup. Always use a center microphone, because most stage action occurs at center stage. Use flanking microphones to cover side areas but observe the 3-to-1 rule and avoid overlapping coverage. Turn up microphones only as needed.

Social events, such as dances, generally require only public address coverage. Use unidirectional, hand-held or stand mounted microphones. Dynamic types are excellent choices, because of their rugged design. The microphone should be equipped with an on-off switch if it is not possible to turn down the microphone channel on the sound system. In any case, turn up the microphone(s) only as needed.

Outdoor use of microphones is, in some ways, less difficult than indoor. Sound outdoors is not reflected by walls and ceilings so that reverberation is not present. Without reflected sound, the potential for feedback is also reduced. However, the elements of nature must be considered: wind, sun, and rain. Because of these factors, dynamic types are most often used, especially in the likelihood of rain. In any case, adequate windscreens are a must. Microphone principles are the same outdoors, so unidirectional patterns are still preferred. Finally, because of frequent long cable runs outdoors, balanced low-impedance models are required.

Conclusion

Though it is one of the smallest links in the audio chain, the microphone is perhaps the most important. As it is the connection between sound source and the sound system, it must interact efficiently with each. Choosing this link successfully requires knowledge of sound and sound systems, microphones, and the actual application. This presentation has included the basic principles not only of microphones but also of sound and sound sytems. Through the examples given, the correct selection and use of microphones for a variety of meeting facility sound requirements has been indicated. Applying these basic principles will assist in many additional situations.

The subject of microphone selection and application for meeting facility sound systems is ever changing, as new needs are found and as microphone designs develop to meet them. However, the basic principles of sound sources, sound sytems, and the microphone that links them remain the same, and should prove useful for any future application.

The successful design of a sound system is not a job for an amateur. It requires extensive technical knowledge and a depth of experience. Shure suggests that if your meeting facility requires a new or updated sound system, contact a reputable sound installation company. Most can design and install a complete sound system to satisfy your requirements.

However, if you need assistance in determining your technical requirements, contact a reputable acoustical consultant in your area. Acting as an "architect of sound", an acoustical consultant will work with you to: 1) understand the needs of your meeting facility; 2) design a sound system that will meet these needs; and 3) make certain that the sound system installer fabricates the system as designed. Though using a consultant will cost more, you will have an experienced advisor on your team that will help you avoid the numerous technical pitfalls of sound systems and acoustics.

Please contact Shure's Applications Group if you need referrals to consultants or sound installation companies in your area.

Design Hints for City Councils, School Boards, and Legislative Chambers

Shure automatic mixers have emerged as the product of choice for city councils, school boards, and legislative bodies. The following are helpful hints on designing and implementing a successful legislative system using a Shure automatic mixer.

Mayor or Chairman Position

- Privacy switches are normally required in legislative chambers. Always use the automatic mixer's MUTE logic terminals to provide this feature. With the Shure AMS automatic mixer, the Off-Attenuation control sets the "depth" of the muting provided when the MUTE terminal is used.
- A visual indicator, like a light emitting diode (LED), showing if a council member's microphone is muted or active is commonly installed where the member can easily see it. This indicator is controlled by the automatic mixer's GATE OUT terminal.
- An all-council mute switch gives the chairman control over every microphone in the system. The automatic mixer's MUTE terminals are employed for this feature.

City Clerk

- The clerk's position should include a switch to mute or activate the microphone on the public lectern. This switch is normally single pole/double throw (SPDT) and is wired to alternately ground the automatic mixer's MUTE terminal or OVERRIDE terminal. This configuration gives complete manual control of the public microphone to the City Clerk. This switch could also be mounted at the mayor's position.

Council Member's Position

- Standard features are a privacy switch using the MUTE terminal and an illuminated visual indication if the microphone is muted or active.

- A gooseneck microphone or stand mounted microphone is recommended. As the microphone is positioned above the table surface, noise from paper shuffling or table tapping is less likely to activate the microphone or be heard through the sound system. A low profile surface mount microphone will always pick up a great deal of unwanted paper noise because the microphone is on the table surface.

- A custom-made rigid tube is an attractive way of mounting a microphone. This eliminates the possibility of a council member repositioning the microphone incorrectly.

- If a low profile surface mount mircophone must be employed, make certain that the rear of the microphone (where the cable exits) is not facing any surface that may reflect sound, like a nameplate or a privacy panel. The reflective surface may interfere with the automatic activation of the microphone or may degrade the acoustic performance of the microphone. [A low profile surface mount microphone may be placed on top of a privacy panel.] If it is absolutely necessary to position a low profile microphone close to a reflective vertical surface, a sound absorbent panel (1" or 2" thick fiberglass covered with open weave fabric) should be installed so that the surface is made as absorptive as possible. This can often improve the inconsistent activation that will may occur if the reflective surface is not treated. Remember, most low profile surface mount microphones are designed to work properly only when the rear of the microphone is facing an open space.

P.A. Feed

- The outputs of the automatic mixer can feed a number of separate devices. The main output generally feeds an equalizer and power amplifier to provide the sound reinforcement needs of the room. If there is an Auxiliary output, it generally feeds a dedicated tape machine which records the council proceedings.

- The automatic mixer's logic terminals are often used to control loudspeaker muting relays for distributed loudspeaker systems. Commercially available duckers and TTL logic-to-relay converters also are useful for loudspeaker muting.

- In very large systems, a "mix-minus" matrix design may be appropriate. In this type of system, a group of microphone signals are fed to all loudspeakers except the loudspeakers closest to the microphones' location. This helps improve control over acoustic feedback.

Recording Applications

- A multi-track tape machine is often used to record the proceedings. One track is often dedicated to document the entire meeting for the public record. The remaining tracks are used to individually record the chairman, the council members, and members of the public whom address the council.

- If it is necessary to record more than one council member microphone on a single track, the Direct Outputs from the automatic mixer can be paralleled to feed that track. Normally, up to twelve Direct Outputs can be connected together.

- A "default"microphone is often employed to maintain a suitable amount of room ambience in the recording. This microphone is activated only if all other microphones are inactive.

Broadcast/Press Feeds

- A distribution amplifier is a useful device to provide six separate feeds from the automatic mixer. Each output of a typical distribution amplifier is switchable for mic or line level providing broadcasters the feed level of their choice. As a distribution amplifer's outputs are isolated from each other, a shorted cable plugged into one output will not affect the other feeds.

- As in Recording Applications, a default microphone should be used to maintain room ambience to the broadcast feeds.

Design Hints for Courtrooms and Video Arraignment Chambers

Shure microphones and automatic mixers have emerged as the product of choice for courtrooms, video arraignment chambers, and other legal facilities. The following are helpful hints on designing and implementing a successful audio system using Shure products.

Suggested Microphones and Their Placement

- A low profile surface mount microphone is often chosen for a courtroom because of its styling. However, this type of microphone should not be placed near any vertical surface that could cause acoustic reflections into the rear of the microphone. This can be difficult to achieve at a lawyer's tables or the judge's bench. For example at the judge's bench, a nameplate or a vertical privacy panel may reflect the talker's voice and cause inconsistent gating or degradation of the microphone's acoustic performance.

- If a low profile surface mount microphone is used at the judge's bench, it should be positioned so that the rear of the microphone is not blocked in any way. Often a small wooden block can be placed under the microphone to raise it higher than a privacy panel. Or the privacy panel can be covered by sound absorbing material (1" or 2" fiberglass covered in porous cloth) to reduce the acoustic reflections.

- Note that because of the low profile microphone's inconspicuous appearance, it is common to have court documents or briefcases accidentally covering the microphone. This will result in very poor audio.

- In place of the low profile surface mount microphone, consider using either a gooseneck mic or the stand mounted mic As these mics are placed above the table or bench surface, paper shuffling or table tapping noise will not be as troublesome.

Judge Position

- A privacy switch is normally required for sidebars. Always use the automatic mixer's MUTE logic terminals to provide this feature. Note that with the Shure AMS automatic mixer, the Off-Attenuation control sets the "depth" of the muting provided when the MUTE terminal is used.

- A visual indicator, like a light emitting diode (LED), showing if the judge's microphone is muted or active is commonly installed where the judge can easily see it. This indicator is controlled by the automatic mixer's GATE OUT terminal.

- An all-mute switch can provide the judge manual control over every microphone in the system. The automatic mixer's MUTE terminals are employed for this feature.

Court Clerk Postion

- The clerk's position should include a switch to mute or activate the microphone for the jury foreman. This switch is normally single pole/double throw (SPDT) and is wired to alternately ground the automatic mixer's MUTE terminal or OVERRIDE terminal. This configuration gives complete manual control of the jury microphone to the City Clerk. This switch could also be mounted at the judge's position.

- A control panel connected to the automatic mixer's logic terminals can be located at the court clerk's position. MUTE and OVERRIDE for each individual microphone are typical controls or an all-mute switch would give the clerk control over every microphone in the system.

- Some clerks are also responsible for video recording of courtroom proceedings. For automatic camera selection, the automatic mixer's GATE OUT terminals can be used to control the Shure AMS880 video switcher interface. The AMS880 will direct a video switcher to call up a certain video camera based on which microphone is activated.

Attorney's Position

- Standard features are a privacy switch using the automatic mixer's MUTE terminal and an illuminated visual indication if the microphone is muted or active.
- A gooseneck microphone or stand mounted microphone is recommended. As the microphone is positioned above the table surface, noise from paper shuffling or table tapping is less likely to activate the microphone or be heard through the sound system. A low profile surface mount microphone will always pick up a great deal of unwanted paper noise because the microphone is on the table surface.
- A custom-made rigid tube is an attractive way of mounting a microphone. This eliminates the possibility of an attorney repositioning the microphone incorrectly.
- If a low profile surface mount microphone must be employed, make certain that the rear of the microphone (where the cable exits) is not facing any surface that may reflect sound, like a nameplate, briefcase, or a privacy panel. The reflective surface may interfere with the automatic activation of the microphone or may degrade the acoustic performance of the microphone. [A low profile surface mount microphone may be placed on top of a privacy panel.] If it is absolutely necessary to position a low profile microphone close to a reflective vertical surface, a sound absorbent panel (1" or 2" thick fiberglass covered with open weave fabric) should be installed so that the surface is made as absorptive as possible. This can often improve the inconsistent activation that will may occur if the reflective surface is not treated. Remember, most low profile surface mount microphones are designed to work properly only when the rear of the microphone is facing an open space.

P.A. Feed

- The outputs of the automatic mixer can feed a number of separate devices. The main output generally feeds an equalizer and power amplifier to provide the sound reinforcement needs of the courtroom. If there is an Auxiliary output, it generally feeds a dedicated tape machine which records the court proceedings.
- The automatic mixer's logic terminals are often used to control loudspeaker muting relays for distributed loudspeaker systems. Commercially available duckers and TTL logic-to-relay converters also are useful for loudspeaker muting.
- In very large courtroom, a "mix-minus" matrix design may be appropriate. In this type of system, a group of microphone signals are fed to all loudspeakers except the loudspeakers closest to the microphones' location. This helps improve control over acoustic feedback.

Recording Applications

- A multi-track tape machine is often used to record the proceedings. One track is often dedicated to document the entire hearing for the public record. The remaining tracks are used to individually record the judge, witnesses, and attorneys.
- If it is necessary to record more than one microphone on a single track, the Direct Outputs from the automatic mixer can be paralleled to feed that track. Normally, up to twelve Direct Outputs can be connected together.
- A "default"microphone is often employed to maintain a suitable amount of courtroom ambient sound in the recording. This microphone is activated only if all other microphones are inactive.

Broadcast/Press Feeds

- A distribution amplifier is a useful device to provide multiple separate feeds from the automatic mixer. Each output of a typical distribution amplifier is switchable for mic or line level providing broadcasters the feed level of their choice. As a distribution amplifer's outputs are isolated from each other, a shorted cable plugged into one output will not affect the other feeds.
- As in Recording Applications, a default microphone should be used to maintain courtroom ambient sound to the broadcast feeds.