Houses of Worship

Audio Systems Guide

Online user guide for House of Worship Audio Systems.
Version: 2.1 (2022-C)

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Houses of Worship Audio Systems Guide

INTRODUCTION

The importance of capturing and reproducing good audio is always critical for houses of worship. Microphones are the first link in the chain - whether it is for live in-person or streaming services, a pre-recorded audio or video message, a daily prayer to a few thousands or even millions at a time. With wireless microphone systems and in ear monitor systems, you can facilitate flexible worship environments, improve performance in challenging acoustic environments, and help create the desired worship experience in modern, media enhanced services.

Purpose of This Guide

The goal of this guide is to provide a solid understanding of several technologies that will be used to create an optimal sound platform for your house of worship. These technologies include:

- **Microphones**- Available in a variety of designs, they are used to pick up the sound source - speech, singing, instruments, etc.
- **Automatic Microphone Mixers and Signal Processors**- Automatic mixers are used to improve speech intelligibility and reduce potential for feedback by activating microphones only when they are addressed. Signal processors are to used to affect frequency response, amplitude, and time properties in microphone output to reduce feedback and noise.
- **Wireless microphone systems**– used to un-tether the speaker and musicians from their fixed spots on the stage without sacrificing any sound clarity.

- **Personal monitoring systems**– used to let the singers and musicians hear the mixes they want at levels that are comfortable to them.
- **Earphones**– used to provide better sound isolation and aesthetics for those that use them.

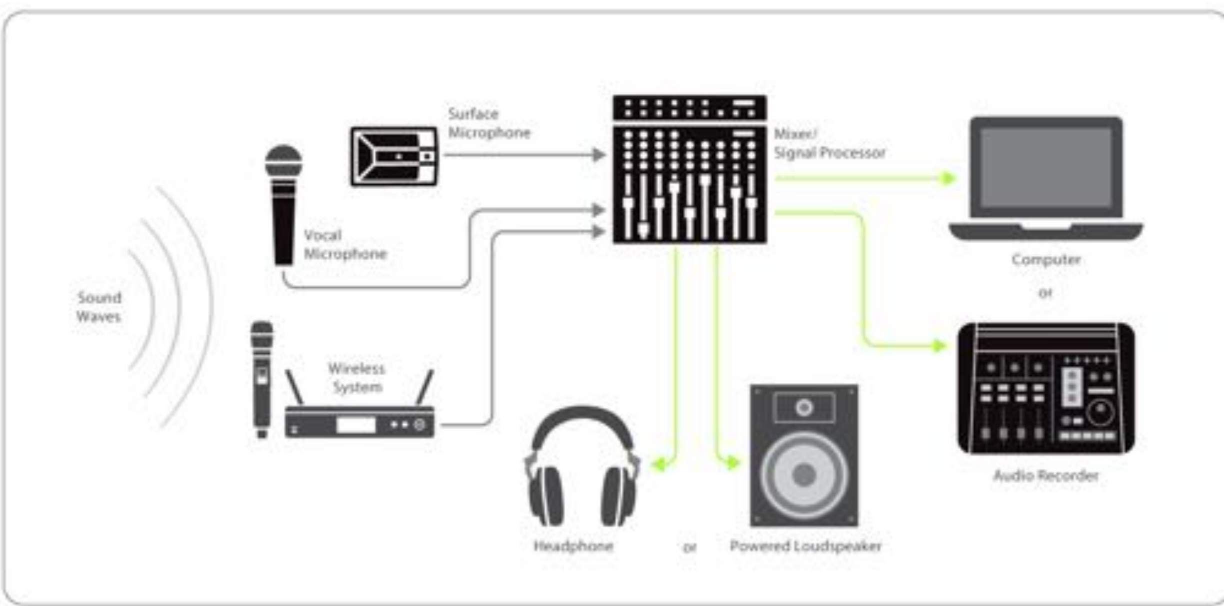
The objective of this guide is to provide the reader with sufficient information to successfully choose and use audio equipment for a variety of house of worship scenarios.

An Introduction to the Sound System

A basic sound reinforcement system consists of several key components:

- Input devices - Usually microphones or instruments, but can also include prerecorded material
- A control device - A mixer allows leveling, processing, and routing of all sound sources picked up by the input devices
- Output devices - For any live performance, the audio is sent to loudspeakers to project the sound reinforcement in room, while the signal can also be sent to additional locations like a recorder, livestream, and/or personal monitor devices.

This arrangement of components is sometimes referred to as the audio chain: each device is linked to the next in a specific order.



Typical Sound System Components

The signal chain starts with input devices. Grey arrows show the input signals and green arrows show the audio output options that depend on your application and requirements.

The primary goal of the sound system in house of worship sound applications is to deliver clear, intelligible speech and high-quality musical sound to the entire congregation. The overall design, and each component of it, must be intelligently thought out, carefully installed, and properly operated to accomplish this goal.

Sound is picked up and converted into an 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. A power amplifier then boosts the line level signal from the mixer to speaker level to drive the loudspeakers. Loudspeakers convert the electrical signal back into sound for the congregation to hear.

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 speaker level (ten Volts or higher). (See The Decibel for more information.)

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

Digital signal processors, such as equalizers, limiters or time delays operate at line level, and may be inserted into the audio chain (usually between the mixer and the power amplifier) or these features may reside within the mixer itself. The general function of these processors is to enhance the sound in some way or to compensate for certain deficiencies in the sound sources or in the room acoustics.

Finally, it may be useful 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 degrade it, sometimes beyond the corrective capabilities of the equipment. In any case, the role of room acoustics in sound system performance cannot be ignored.

What is Good Sound?

There are three primary measures of sound quality:

- Fidelity
- Intelligibility
- Loudness

In a house of worship the quality of sound will depend on the quality of the sound sources, the sound system, and the room acoustics.

Fidelity

The fidelity of sound is primarily determined by the overall frequency response of the sound arriving at the listener’s ear. It must have sufficient frequency range and uniformity to produce realistic and accurate speech and music. All parts of the audio chain contribute to this: a limitation in any individual component will limit the fidelity of the entire system. Frequency range of the human voice is about 100-12kHz, while a compact disc has a range of 20-20kHz. A telephone has a frequency range of about 300-3kHz and though this may be adequate for conversational speech, it would certainly be unacceptable for a sound system. However, even a high fidelity source reproduced through a high fidelity sound system may suffer due to room acoustics that cause severe frequency imbalances such as standing waves.

Intelligibility

The intelligibility of sound is determined by the overall signal-to-noise ratio and the direct-to-ambient sound ratio at the microphone and the listener’s ear. In a house of worship, the primary signal is the spoken word. The noise is the ambient noise in the room as well as any electrical noise added by the sound system. In order to understand speech with maximum intelligibility and minimum effort, the speech level should be *at least* 20 dB louder than the noise at every listener’s ear. To ensure this, you need to try to achieve a speech-to-noise ratio of at least 30 dB. This means that – at the microphone – the voice should be 30 dB louder than the ambient noise.

If the speech is not sufficiently louder than the ambient noise at the microphone, there are only a few possible remedies:

1. Have the talker speak louder. For non-professional speakers this is usually not feasible.
2. Make the room quieter. This is typically impossible without significant modifications to the building's structure, interior decor, and/or air handling system.
3. Move the microphone closer to the talker's mouth.

Moving the microphone closer may require switching to a different style of microphone. If a lavalier microphone does not deliver intelligible speech without excessive noise or feedback, for example, the only practical alternative may be to have the talker wear a headworn microphone or use a handheld mic.

Loudness

The loudness of the speech or music at the farthest listener must be sufficient to achieve comfortable levels for speech and perhaps more powerful levels for certain types of music. These levels should be attainable without distortion or feedback.

Potential Acoustic Gain (PAG) is a measure of how much gain or amplification a sound system will provide before feedback occurs. This turns out to depend very little on the type of system components, but very much on **the relative locations of microphones, loudspeakers, talkers, and listeners**. (See Potential Acoustic Gain in this guide for more information. Or, visit our website to use our free PAG NAG calculator based on the details of your installation.)

The Sound Source

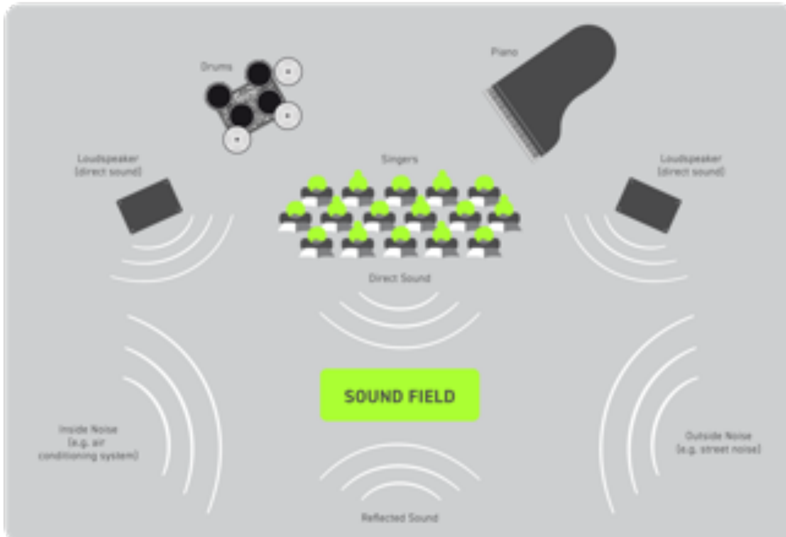
The sound sources most often found in worship facilities are the speaking voice, the singing voice, and a variety of musical instruments. Because spoken word is such a critical aspect of the service, special attention is given to microphone selection and placement, which often requires wearing the microphone for best results.

An Important Sound Source

The pastor's audio quality is one of the most important considerations when designing the audio system.

In addition to these desired sound sources, there are certain undesired sound sources that may be present: building noise from air conditioning or buzzing light fixtures, noise from the congregation, sounds from street or air traffic, etc. Even some desired sounds may become a problem, such as an organ that overpowers the choir.

The loudspeakers of the sound system can become an undesired source for microphone pickup, causing feedback (an annoying howl or ringing sound) if microphones “hear” too much of the loudspeakers.



The Sound Field: Desired and Undesired Sources

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 congregation. The acoustic nature of an area may have a positive or a negative effect on the sound produced by voices, instruments, and loudspeakers before it is picked up by microphones or heard by listeners: absorbing or diminishing some sounds while reflecting or reinforcing other sounds.

Microphones

Microphone Characteristics

The microphone is the first link in the audio chain and is therefore critical to the overall performance of the sound system. Proper selection of microphones depends on an understanding of basic microphone characteristics and on a knowledge of the intended application.

There are five areas of microphone characteristics that should be considered when selecting a microphone for a particular application. They are:

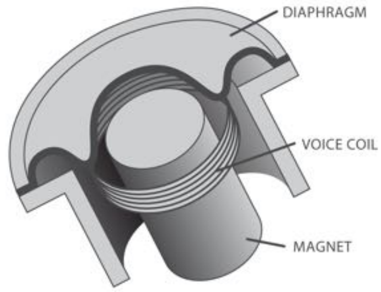
- **Operating Principle**: How does the microphone change sound into an electrical signal?
- **Frequency Response**: How does the microphone sound?
- **Directionality**: How does the microphone respond to sound from different directions?
- **Electrical Output:** How does the microphone output match the sound system input?
- **Physical Design**: How does the design relate to the intended application?

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

The operating principle describes the type 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 picks up sound and converts it into an electrical signal. The operating principle determines some of the basic capabilities of the microphone.

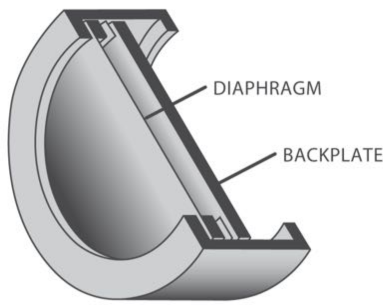
The two most common types are dynamic and condenser. Although there are other operating principles used in microphones (such as ribbon, crystal, carbon, etc.) these are rarely encountered in worship facility sound applications.

Dynamic microphones employ a diaphragm/voice coil/magnet assembly which forms a miniature sound driven electrical generator. The motion of the voice coil in the magnetic field generates an electrical signal that corresponds to the sound.



Dynamic microphones are mechanically more simple and rugged, can operate in extremes of temperature or humidity, and can handle the highest sound pressure levels without overload. Although they cannot be made very small, dynamic microphones are the most widely used for both vocal and instrument applications in worship facilities.

Condenser microphones are based on an electrically-charged diaphragm/back plate assembly which forms a sound-sensitive capacitor. Sound waves cause the diaphragm to move, which varies the spacing between the diaphragm and back plate, which produces an electrical signal corresponding to the sound.



All condenser microphones contain additional circuitry that requires power either from batteries or from “phantom” power that is supplied through the microphone cable. Note that the electronics produce a small amount of noise (hiss), and that there is a limit to the maximum signal level that the electronics can handle without causing distortion. Well-designed condenser microphones, however, have very low noise levels and can also tolerate high signal levels.

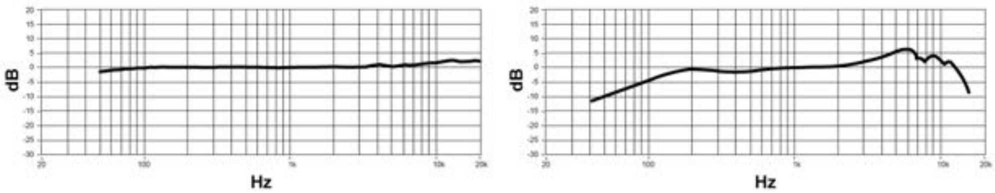
Condenser microphones are more sensitive than dynamics, and have extended high frequency response that adds detail to voices and instruments. They can also be made very small, making them ideal for lavalier, headworn, or instrument-mounts where tight space or concealment is needed.

The decision to use a condenser or dynamic microphone depends not only on the sound source and the signal destination but on the environment as well. From a practical standpoint, if the microphone will be used in a high-impact environment such as a community shared space or for outdoor sound, a dynamic microphone would be a good choice. In a more controlled environment, for example, in a sanctuary, auditorium, or theatrical setting, a condenser microphone might be preferred for some sound sources, especially when the highest sound quality is desired.

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 characteristic “sound” of the microphone.

The two general types of frequency response are *flat* and *shaped*. These terms refer to the general shape of the graphical frequency response curve.



A flat response on the left compared to a shaped response on the right

A microphone that provides the same output at every audible frequency has a frequency response graph that is a generally flat line, and is said to have a flat response. This means that the microphone reproduces sound with little or no variation from the original source. Flat response microphones typically have an extended frequency range and can reproduce very high and/or very low frequencies as well. Wide-range, flat response microphones have a natural, high-fidelity, “uncolored” sound.

When a microphone is more sensitive to certain frequency ranges than others, it has a shaped response. A shaped response mic is more sensitive to important or desirable frequency ranges and less sensitive to undesirable ones. The frequency graph appears as a varying line with specific peaks and dips.

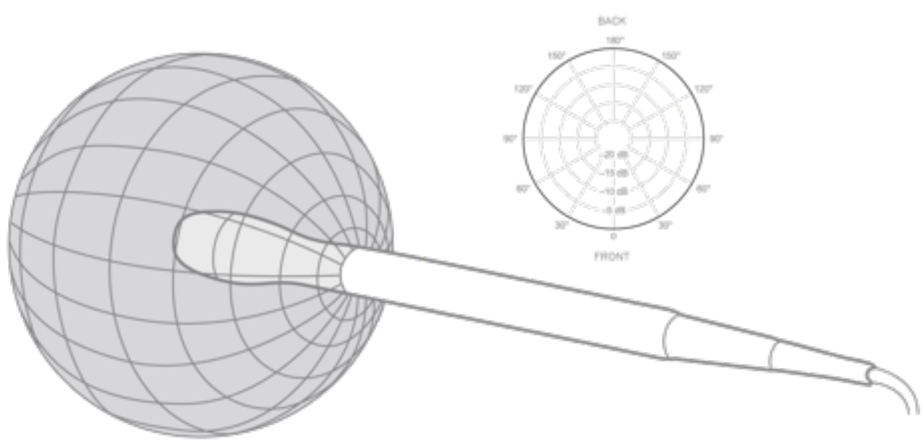
Normally, microphones with flat, wide-range response are recommended for pickup of acoustic instruments, choral groups and orchestras, especially when they must be placed at some distance from the sound source. Mics with shaped response are especially good for vocal use. The frequency response includes enhanced sensitivity to the upper mid-range (often called a "presence rise") for speech intelligibility and vocal clarity, and reduced sensitivity to low frequencies to reduce pickup of room noise and mechanical vibration.

The frequency response of some microphones is adjustable to tailor the microphone to different applications. Most common are low-frequency roll off controls, which can help prevent “rumble”, and presence rise switches to enhance intelligibility.

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

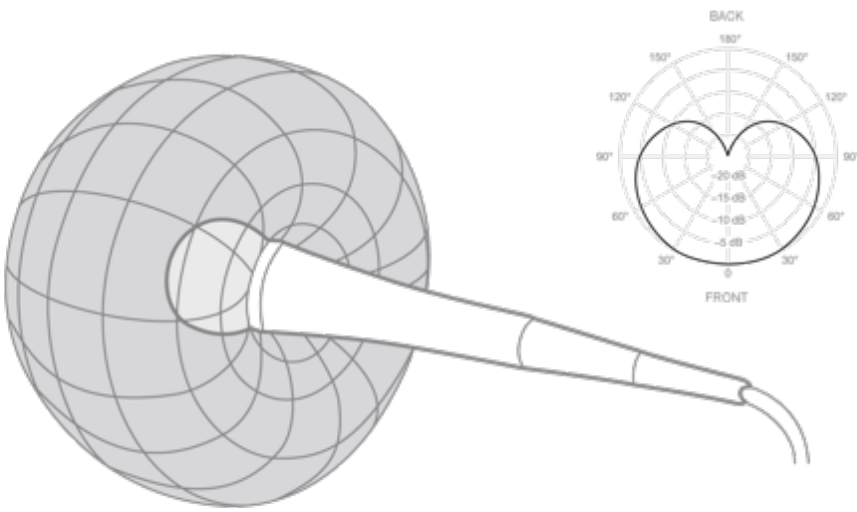
The directionality of a microphone is defined as the variation of its output when it is oriented at different angles to the direction of the sound. It determines how best to place the microphone relative to the sound source(s) in order to enhance pickup of desired sound and to minimize pickup of undesired 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 has a polar graph that is 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 pick up sound from a wide area, but cannot be “aimed” to favor one sound over another.



Omnidirectional Microphone

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, because of its heart-shaped polar pattern.

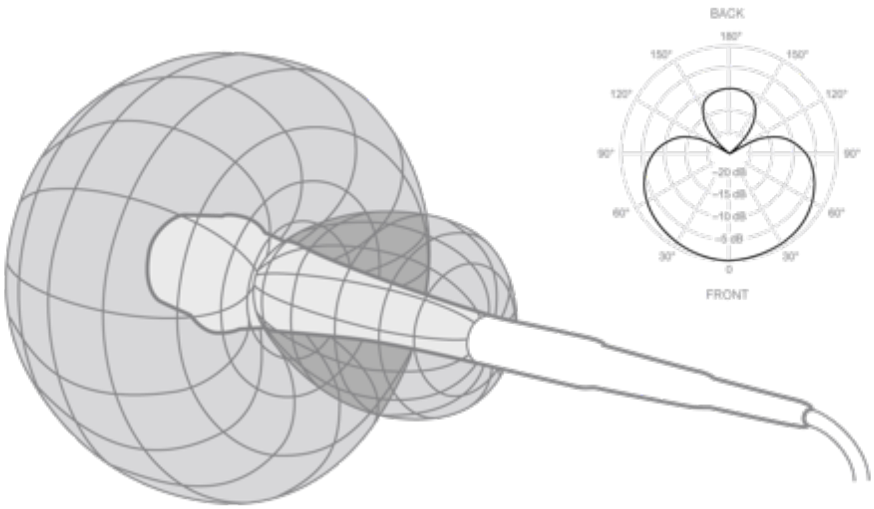


Cardioid Microphone

A cardioid type is most sensitive to sound coming from in front of the microphone (the bottom of the “heart”). On the polar graph this is at 0 degrees, or “on axis”. It is less sensitive to sound from the sides (“off-axis”), and the direction of least sensitivity is toward the rear (the notch at the top of the “heart”). 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 a desired, direct sound by orienting its axis toward the sound. It may also be aimed away from an undesired, direct sound by orienting its null angle toward the sound. In addition, a unidirectional microphone picks up less 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.

Unlike the cardioid, other unidirectional microphones 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 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 null angle.



Supercardioid Microphone

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 least overall 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.

CHARACTERISTIC	ONMI-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°
REAR REJECTION (relative to front)	0	25 dB	12 dB	6 dB	0
AMBIENT SOUND SENSITIVITY (relative to omni)	100%	33%	27%	25%	33%
DISTANCE FACTOR (relative to omni)	1	1.7	1.9	2	1.7

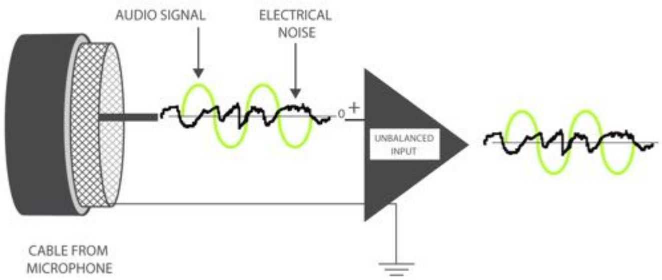
Comparison Chart - Polar Patterns and Directionality

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 configuration. 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.

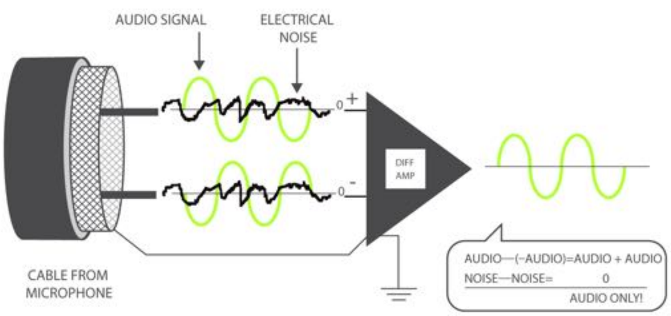
The *sensitivity* of a microphone is defined as its electrical output level for a certain input sound level. *Impedance* is essentially the electrical resistance of the microphone's output signal: 150-600 ohms for low impedance (low Z), 10,000 ohms or more for high impedance (high Z).

The output configuration can be either *balanced* or *unbalanced*. A balanced output carries the signal on two conductors that are wrapped with a metallic shield. The signals on each conductor are the same level but opposite polarity (one signal is positive when the other is negative). An unbalanced output signal is carried on a single conductor (plus shield) and is common for instruments such as electric/acoustic guitars, keyboards and others.



Unbalanced Inputs

Balanced, low impedance microphone connections (typically with XLR connectors) are the standard for all professional microphones because they allow for long cable runs (over 100 feet) with no loss of quality while reducing pickup of electrical noise. Audio mixers commonly include balanced mic inputs (usually with phantom power for condenser microphones).



Balanced Inputs

PhysicalDesign: How does the design relate to the intended application?

Commonly used microphones for house of worship sound applications include handheld, user-worn, gooseneck, and boundary. 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 and singing. Since they are usually held (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, since they are often used very close to the mouth, they require a “pop” filter or windscreen to minimize explosive breath sounds. Size, weight and feel are also important considerations for a handheld microphone.

Handheld Vocal

- **Headset and Lavalier Microphones** include lavalier or “lapel” types that may be attached directly to clothing, and head-worn models. In particular, headworn microphones have become much more common as their size has decreased. The proximity of the headworn microphone to the mouth results in much better sound quality and vastly increased gain-before-feedback when compared to a lapel type. Small size and unobtrusive appearance are the critical characteristics for user-worn microphones.

Headset and Lavalier Microphones

- **Gooseneck** (or gooseneck-mounted) microphones range from full size microphones on heavy-duty flexible stands to miniature types on slim flexible goosenecks Gooseneck microphones can be permanently attached to a podium or lectern or may be mounted on a base (wired or wireless) that can be moved as needed. Shock isolation is essential to prevent pickup of vibration from the mounting surface. Windscreens are necessary for close-up vocals or if used outdoors, especially with condenser types.

Gooseneck

- ***Overhead*** microphones are typically miniature condenser types suspended from the ceiling over a choir or ensemble. Overhead microphones require careful aiming and positioning to provide even coverage, and their distance from the source limits the gain-before-feedback that can be achieved.

Overhead

- **Boundary** microphones are also used in fixed positions, but the surface to which they are attached is essential to the operation of the microphone. Boundary mics are usually mounted on altars, tables, or floors to cover a certain area. They depend to a great extent on the size and reflective properties of the mounting surface for their frequency response and directionality. However, they offer a very low profile and can minimize certain acoustic problems due to reflected sound. Appearance and physical environment play an important part in the selection of boundary microphones.

Boundary

It should be noted that almost any combination of the other four 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 principle, frequency response, and directional pattern.

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 characteristics must be properly specified to yield the best results.

Automatic Mixers and Signal Processors

There are several tools available to achieve better sound in the room. These include automatic microphone mixers to control microphone levels, and signal processors to improve the audio quality (various types of equalizers) and reduce unwanted effects (such as feedback and comb filtering).

Automatic Mixers

An automatic microphone mixer should be considered for speech applications, whenever a small system with multiple microphones (four or more) are used. This is particularly true if the sound system is intended to run hands-free, that is, without a live operator.

Examples of Automatic Microphone Mixers (shown front and back)

The reasons for using an automatic microphone mixer relate to the behavior of multiple microphone systems. Each time the number of open or active microphones increases, the overall system gain or volume also increases, as if the master volume control were turned up. **As more microphones are added, the potential for feedback grows.**

In addition, unwanted background noise increases at the same time. Here, the effect is a loss of intelligibility as the background (ambient) noise level rises closer to the level of the desired sound. (See Potential Acoustic Gain)

The solution when using microphones for speech is to automatically 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 volume must be reduced appropriately to prevent feedback and ensure minimum noise pickup.

In some systems, ordinary microphones are used and all of the control is provided by the hardware or software mixer. In others, microphones with additional features are integrated with the mixer to provide enhanced control.

There are several techniques used to accomplish channel activation or gating in an automatic mixer system. In most systems, a microphone is gated on when the sound that 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. In any case, the threshold should be set so that the microphone is not activated by background noise but will be activated by normal sound levels.

Some recent automatic mixers incorporate digital signal processing (DSP) or noise adaptive threshold circuitry. These have the ability to distinguish steady signals such as background noise from rapidly changing signals like speech. They can automatically and continuously adjust individual channel thresholds as ambient noise conditions change. In addition, some designs can recognize that the same signal is being picked up by more than one microphone. In that case, only the channel with the strongest signal is activated. This prevents both microphones from being activated when a talker is in between two microphones for example.

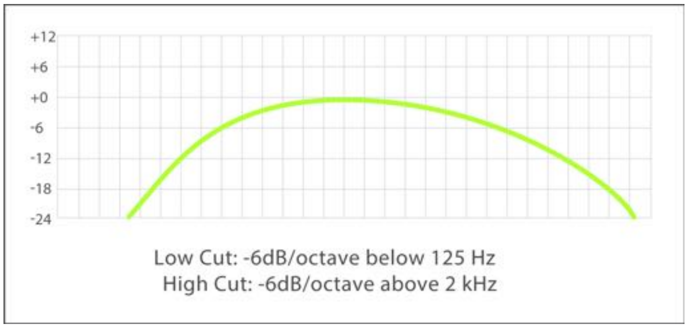
Many automatic microphone mixers have additional control circuitry, often in the form of logic connections. These are electrical terminals that can be used for a variety of functions, including: microphone status indicators, mute switches, loudspeaker attenuation, and the selection of “priority” channels. Some automatic mixers have an adjustable “off attenuation” control: instead of gating the microphone completely off, it can be “attenuated” or turned down by some finite amount, to make 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 the microphone off between words or short pauses. In addition, a function which locks on the last microphone activated ensures that at least one microphone is on, even if no one is speaking. Finally, most automatic mixing systems are able to be expanded by adding individual channels and/or by linking multiple mixers together to control large numbers of microphones simultaneously.

Signal Processors: Equalizers and Feedback Control

Equalizers

Signal processors fall into three main categories based on which property of the audio signal they affect: equalizers affect frequency response, dynamics controllers affect amplitude, and delays affect time properties such as phase. Each of these can be useful in the operation of microphones but equalizers are of particular interest because of their potential use in feedback control.

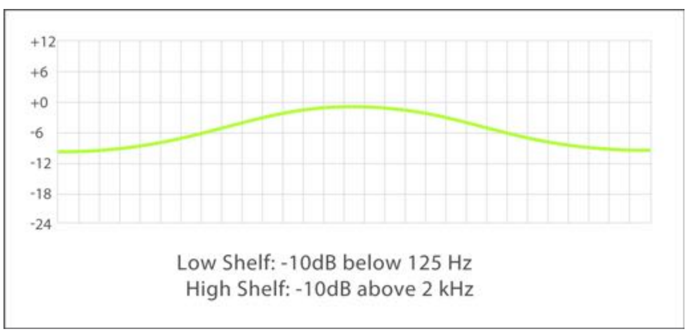
Feedback is frequency-dependent. Because it occurs first at peaks in the overall sound system frequency response, equalization of the response may significantly affect the onset of feedback. System response peaks may be due to many factors including system components, transducer location, or room acoustics. In principle, the system response must be reduced at those frequencies which trigger feedback. The goal is to allow the system to operate at higher gain without ringing or feedback.



Low Cut and High Cut Filters

Equalizers are frequency-dependent filters that fall into several categories based on the characteristics of the filters and their adjustment. Hi-cut and lo-cut (or, alternately, lo-pass and hi-pass) filters progressively attenuate or reduce all frequencies above (or below) a certain cutoff frequency. That is, the attenuation increases with frequency further above (or below) the cutoff frequency. The cutoff frequency may be adjustable: down to 5000 Hz for hi-cut and up to 500 Hz for lo-cut. The “slope” or rate of attenuation may also be adjustable from a minimum of 6dB/octave to as steep as 24dB/octave. Hi-cut and lo-cut filters are used to reduce the bandwidth or frequency range of the signal to remove unwanted high frequency or low frequency sounds such as hiss or rumble.

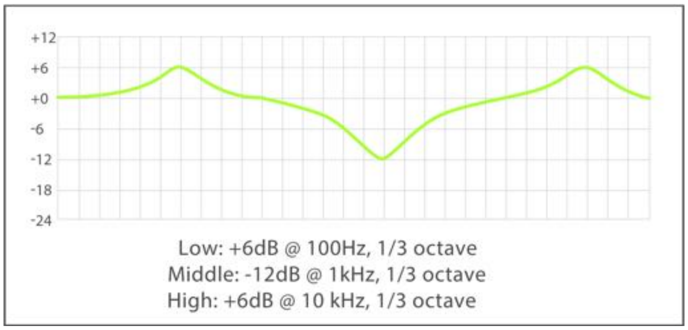
Shelving equalizers allow low frequencies (or high frequencies) to be cut or to be boosted. The cut or boost is not progressive: it is the same at all frequencies below (or above) the filter frequency. The response curve looks somewhat like a shelf above or below the filter frequency. The amount of cut or boost is adjustable typically up to ± 15dB. The filter frequency is usually fixed: about 250 Hz and below for low frequencies, about 8000 Hz and above for high frequencies. Shelving equalizers are used for general response shaping at low and high frequencies. They are the type of filter used as “bass” or “treble” tone controls.



Shelving Equalizers

Bandpass equalizers allow frequencies within a certain band or range to be cut or boosted. They are classified according to their bandwidth and/or according to the number of filters employed. Bandwidth is usually given as a fraction of an octave (an octave represents a doubling of frequency such as 400 Hz-800 Hz or 4000 Hz-8000 Hz). For example, a mid-range tone control is a single bandpass filter with a 1 octave bandwidth designed to affect the frequency range between a bass control and a treble control, typically 500 Hz-1000 Hz. Again, the range of cut or boost is typically up to ± 15 dB. Bandpass filters have fixed frequency and fixed bandwidth.

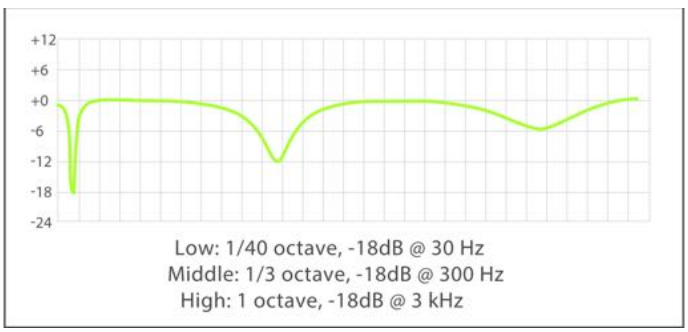
Sets of multiple bandpass filters are used for more precise overall response shaping. When vertical slide controls are used for adjustment these are called graphic equalizers because the shape of the resulting response curve is visually approximated by the control positions. Graphic equalizers also have fixed frequency and fixed bandwidth. Typical variations are 1 octave (8-10 bands), 1/2 octave (12-15 bands), and 1/3 octave (27-31 bands). The narrower the bandwidth, the more filters are available and the more precise the adjustment capability.



Graphic Equalizer

A set of bandpass filters whose frequency and bandwidth can also be adjusted is called a parametric equalizer because all of its “parameters” are adjustable. Parametric equalizers can be “tuned” to any desired frequency, adjusted to a suitable bandwidth, and boost or cut as needed. They typically have a frequency range of 20-20,000 Hz, a bandwidth range of 1/10 to 2 octaves, and cut or boost of ± 15 dB. Most parametric equalizers have at least 3-5 independent filters, though some midrange controls on mixing consoles are actually a parametric filter. Parametric equalizers can provide very precise frequency response shaping.

A special type of parametric filter is the “notch” filter. It has both variable frequency and bandwidth but is used in a “cut only” mode, typically down to -18 dB. In addition, the bandwidth of some notch filters can be as narrow as 1/40 octave. Notch filters are the most useful filters for feedback control because they allow precise attenuation at any frequency with minimal effect on adjacent frequencies. A number of notch filters can be activated with very little audible effect on overall sound quality.

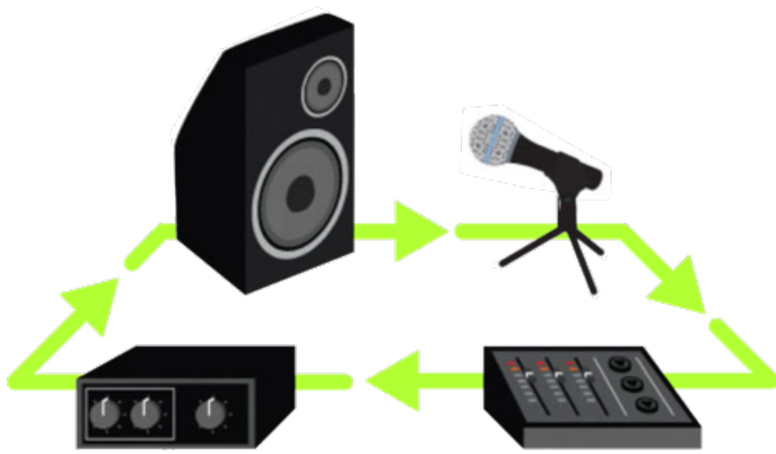


Parametric Equalizers

Other equalizer types, even 1/3 octave graphics, have a very noticeable effect on sound quality due to the relatively large bandwidth of their filters, especially when adjacent filters are used to reduce an “in between” frequency. Similarly, use of hi-cut, lo-cut, or shelving equalizers for feedback control can result in severe loss of sound quality and is warranted only if the feedback is at an extremely high or low frequency.

Feedback Control

The use of an equalizer for feedback control is of course limited to the degree that the feedback is due to inequalities in system components or room acoustics. It cannot compensate for badly-located microphones and/or loudspeakers and certainly will not eliminate all possibility of feedback. Poorly-designed systems or unreasonable operating conditions can’t be fixed by even the most powerful equalizer. Nevertheless, judicious use of equalization can improve the feedback stability of a well-designed system and may even allow a marginal system to operate adequately.



Feedback Loop

The traditional approach to “ringing out” or equalizing a sound system for feedback problems is to gradually bring up the gain of the system until ringing or feedback begins, identify the offending frequency, and insert an appropriate filter until the feedback stops. The process is repeated until either the desired gain is reached (hopefully) or all the filters are used up. The most difficult steps are: identifying the feedback frequency and inserting the appropriate filter. Even very experienced sound engineers often have to rely on special equipment to pinpoint the feedback frequency. In addition, the use of parametric filters or notch filters is not very intuitive.

Feedback Controllers

Feedback controllers automatically identify and reduce feedback using complex algorithms (mathematical modeling techniques) to identify sustained single frequency sounds and to deploy a notch filter of the correct frequency and attenuation. These devices typically have 5-10 filters that can be automatically set. The filters are narrow enough (1/10 octave) that their effect is not noticeable beyond the reduction of feedback. A bypass switch is usually provided to compare the equalized and unequalized sound after the filters have been set.

Some feedback controllers have other functions built-in. These may include other types of equalizers such as graphic or parametric, or other types of processors altogether such as limiters and time delays. Certain models offer computer interfaces for programming, external control, and monitoring.

Though none of these devices can anticipate feedback, they can still respond to the onset of feedback or ringing more quickly and accurately than most human operators. However, feedback controllers do not equalize the system for good sound, merely for least feedback. It is still up to the system designer and operator to ensure the desired sound quality. Within the limitations mentioned earlier, such automatic feedback controllers can be quite useful. They can be used on the main sound system, the monitor system, or even inserted on an individual channel. If the sound system is normally controlled by an operator, they can assist in the ringing out process. The operator merely continues to slowly turn up the system level until the major feedback frequencies have been identified and “notched” out. Alternately, the device can be left active to take care of feedback that may occur during unattended system operation. However, these devices cannot distinguish between sustained musical notes and feedback. That is, a sustained note on a keyboard or guitar may be interpreted as feedback and a corresponding filter will be inserted at that frequency. For this reason, it is recommended that for musical performances these devices should be “locked” after the initial ringing out. When used properly, feedback controllers can improve gain-before-feedback by up to 6-10 dB. Remember that more substantial improvements can often be made just by repositioning microphones or loud speakers.

Wireless Microphone Systems

Benefits of Going Wireless

A wireless microphone system offers undeniable advantages to pastors, speakers, singers, and worship bands. Using a wireless microphone system lets you take advantage of an easy setup workflow and eliminates unnecessary cable clutter in the House of Worship. And most importantly, it allows the pastor and vocalists to move throughout the sanctuary while performing or speaking freely.

Houses of worship have unique audio challenges and needs that are addressed by wireless systems. This includes the configuration of the space itself, as well as the various expectations and desires of the worship team and the worshipers.

Let’s look closer at these two main advantages:

1. Cable-free mobility for the pastor, worship leader and worship musicians
2. Fewer cables, which provides a cleaner, less cumbersome worship space

- **Greater mobility** – As praise bands become more elaborate and the congregations’ expectations of more interaction increases, other musicians, such as the horn player and the guitarist, are finding that the cable on the wired microphone is limiting their ability to bring their worship closer to – and often into – the congregation.

  Additionally, the pastor might want to lend a voice to the praise band. With a wireless microphone, he or she can simply walk across the platform and join in.

- **A cleaner worship space** – Again, as praise bands become more elaborate, as more and more guest speakers are added to the platform, the number of people who need to be miked increases. This results in the need for more and more microphone cables and stands.

  Wireless systems eliminate the cables on the platform and allow new presenters and musicians to join the celebration without adding yet another cable to the clutter.

  For example: you want to feature a member of the choir in the song. Simply hand her the pre-set wireless microphone and she can walk forward on the platform and add her voice to the worship without adding another cable to the stage.

  Then, when her part is over, she can hand the microphone to the next featured singer or step back and rejoin the choir.

In addition, there are two more reasons to consider upgrading your sound platform to include these technologies: *hearing conservation and vocal strain*. There has been a great deal of research lately on the hearing loss of people who are constantly exposed to sound, even if the sound is not always overly loud. There has also been more understanding of the vocal strain caused by having to continually sing over high volume. Since worship team members are often part of multiple services weekly, if not daily, these two reasons, alone, would merit considering personal monitors and earphones for your services.

All in all, the benefits of including wireless microphone systems and personal monitors into your house of worship will likely more than pay for themselves in the added richness of the overall sound for your congregation and the increased control for those who use them.

Digital Wireless Systems: The Digital Advantage

Recent advancements in digital wireless microphone technology and design have given the user of wireless microphones access to better sounding, better performing systems. This digital technology has positives and negatives.

Enhancements

- **More natural wired-like sound** - The limitations of analog wireless required a compression stage in the transmitters and expanders in the receivers. In digital systems there is no need for this. Digital wireless systems can preserve full dynamic range and audio quality without the subtle but unwanted changes to the original sound of the microphone or guitar.
- **More simultaneous systems** - When implemented properly, digital wireless systems create less problems and conflicts with other wireless systems. If you anticipate the need for significant expansion of your wireless mic/instrument use, digital systems are worth considering.

Considerations

- **Latency** - Because it takes a little time to make the analog to digital (and subsequent digital to analog) conversion in digital wireless systems, there is always some inherent delay or “latency” in the signal as it gets processed, transmitted, and processed again. Too much of this latency is distracting to the user and can create timing problems with musicians.
- **Shielding** - Some older and not well shielded mics and pickups may be noisier with digital systems when compared to analog systems. Make certain that the microphones and pickups are properly shielded for use with modern, digital equipment.

The Basics of Any Wireless Microphone System

Let's start with the basics of any wireless microphone system.

Wireless Components

1. The Microphone

As previously discussed, a variety of microphone designs are available to match the sound source and application. Common microphone types for wireless systems include:

- Handheld microphone (integrated with the transmitter base)
- Headworn vocal microphone
- Lavalier (lapel) vocal microphone
- Clip-on instrument mic
- Direct input from a guitar/bass (which replaces the microphone via a cable)

2. The Transmitter

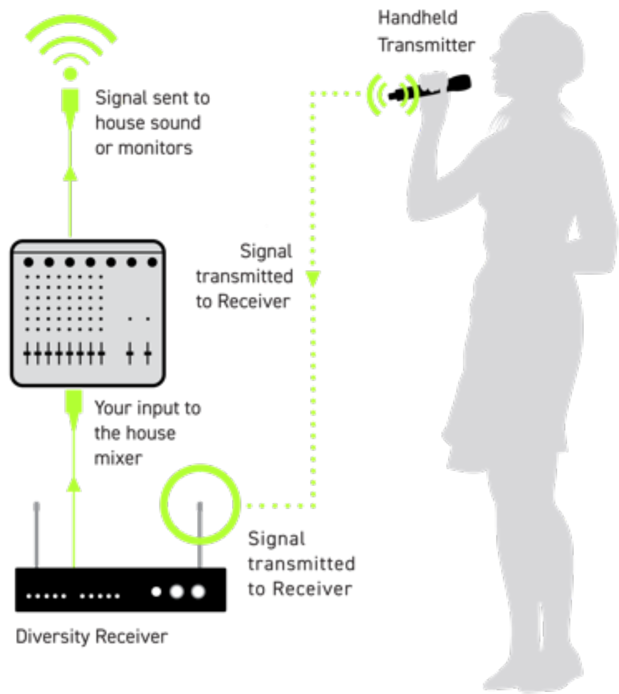
The wireless transmitter is typically built into the base of the handheld microphone or as a separate bodypack device that clips onto the belt or clothing. Its function is to convert the audio signal from the microphone to a radio signal and send this signal to the receiver.

Important to note: These radio signals are sent from the transmitter to the receiver on a predetermined radio frequency – in the same way your local radio and television stations transmit their broadcasts.

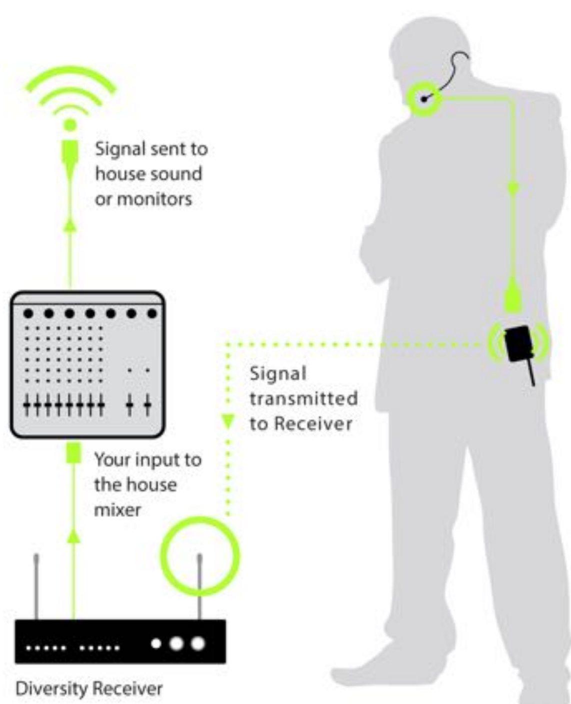
3. The Receiver

The receiver operates on the same channel as the transmitter. It receives the radio waves from transmitter and converts them back to an audio signal. The audio output from the receiver is usually the same as what would come from a regular wired microphone, and so you would plug it into the sound system the same way you would plug in a microphone.

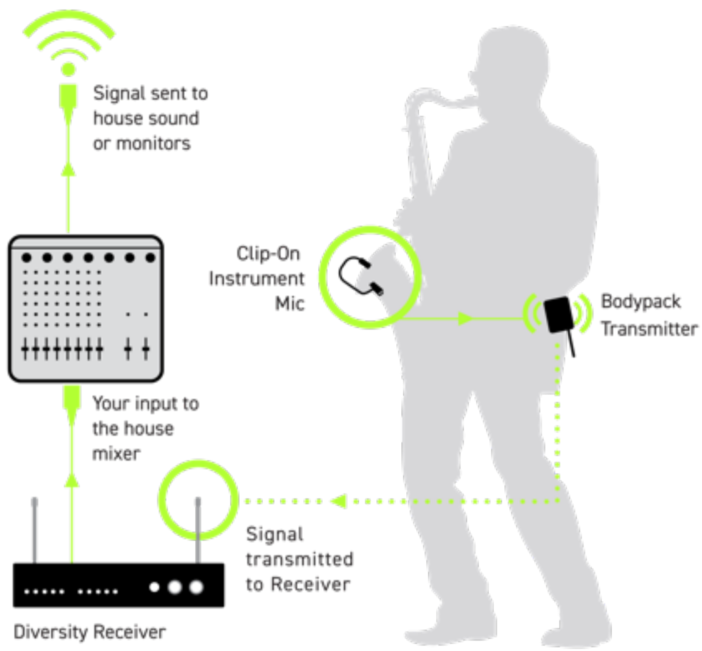
Examples of Wireless Systems



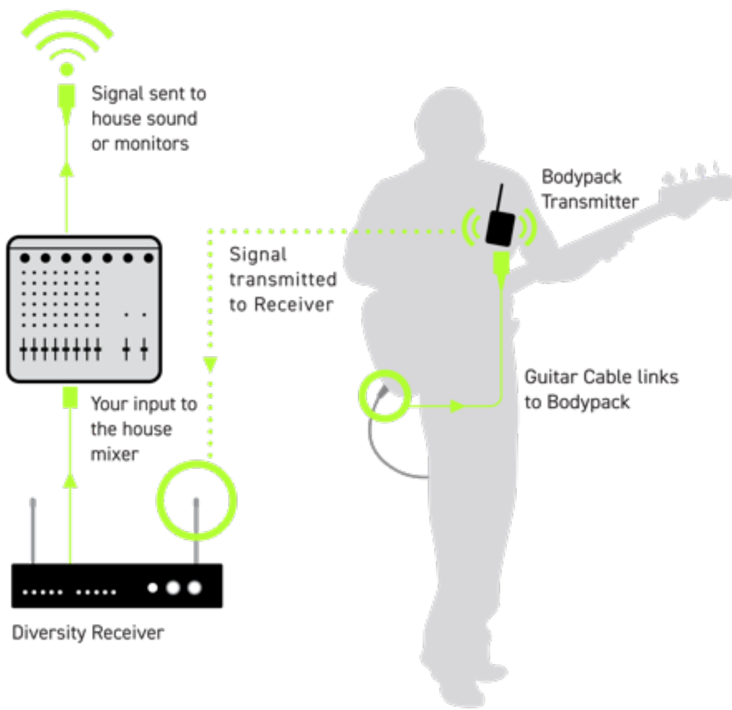
Handheld Wireless Microphone



Headworn Wireless Microphone



Instrument with Wireless Transmitter



Guitar with Wireless Transmitter

Key Concepts for a Successful Wireless Setup

Each Channel Operates on a Single, Clear Frequency

Most professional wireless microphone systems transmit and receive sound on a specific radio frequency. These frequencies are mainly grouped into two large bands, or ranges: VHF and UHF. The most popular band is UHF (Ultra High Frequency) band of 470 - 960* MHz, while some systems work in the VHF (Very High Frequency) band of 174 – 216* MHz.

The UHF band contains several sections that are readily available for wireless microphone systems. In the UHF band, you can achieve very high-quality audio and accommodate many concurrent systems. This band has been the standard for many years and typically gives the strongest and best performance due to its extensive frequency range and smaller antennas.

By contrast, you can find several wireless microphones systems that operate in the 2.4 GHz band. These systems use frequency sharing technology that don't operate on a single frequency per channel. Keep in mind that the 2.4 GHz frequency band is the same as your computer Wi-Fi, which makes its operating range much shorter. It is best practice with 2.4 GHz wireless systems to make sure that you have a controlled environment.

**Region dependent*

Here are some simple guidelines related to frequencies:

- Each wireless system must be on a different frequency.
- Most wireless microphones share the same frequencies used by TV stations, both VHF and UHF. Since TV stations are much more powerful than wireless microphones – and since the Federal Communications Commission (FCC) requires you to do so – you need to avoid local TV channels.
- You also have to avoid frequencies that are already used within your house of worship or those in use by other organizations nearby.

- Most manufacturers have online tools to help you select the best range based on your model and location. They can also help select the right frequencies when multiple systems are used.

Receiver and antenna placement

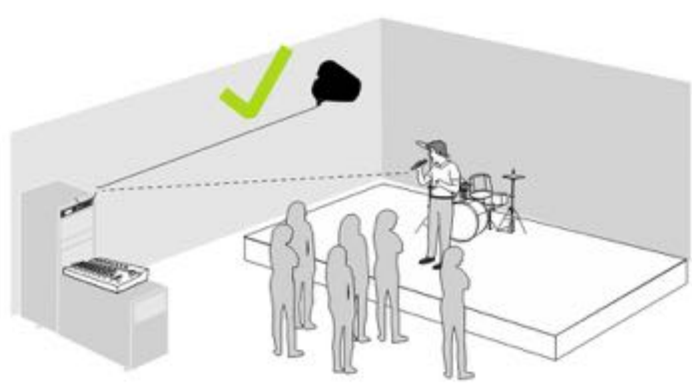
Wireless microphone systems include antennas on both the receiver and transmitter.

Antennas range in shape, size and even quantity. Some can be obvious; such as on bodypack transmitters while others are located internally; such as for many handheld transmitters.

Here, again, the discussion can quickly become technical, so we have outlined a few basic principles to help you avoid interference and increase the likelihood you will get clear audio.

Proper and improper antenna positions:

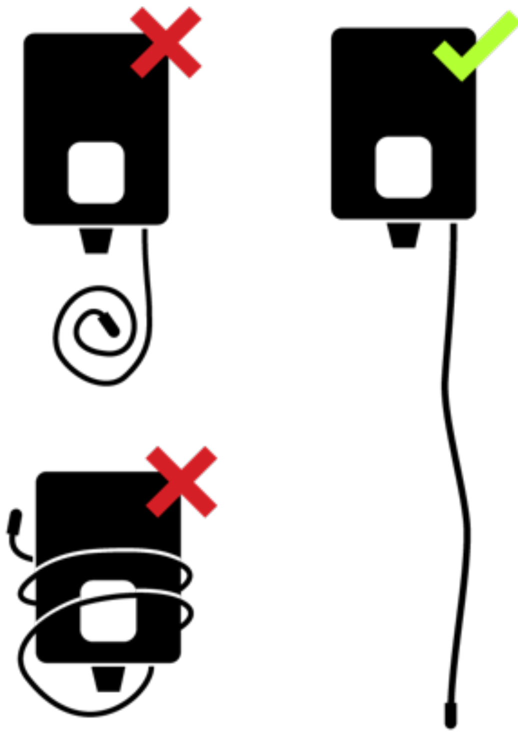
- To provide optimal communication between the transmitter and receiver, you must maintain a clear line of sight between the transmitter and the receiver antennas.



Maintain Line of Sign

This drawing shows two options for antennas. If the line of sight from the antenna is obstructed by the crowd or other materials (shown by dotted line), then choose to remote-mount the antennas. The remote antennas provide a direct line of sight from the performance area to the receiver.

- Remote or receiver antennas should be placed above the congregation or other obstructions so the transmitter and the receiving antenna have line of sight.
- Antennas should always be kept as clear as possible from obstructive surfaces or metal materials. Never curl up the antenna into a pocket, or wrap it around the bodypack.



- Never let antennas touch one another.

When mounting receivers onto racks, use remote mounting to maintain line-of-sight to the transmitters. Here are some additional guidelines:

- Use two antennas when remote mounting - diversity receivers provide superior performance and guards against dropouts.
- For locations where a great number of wireless microphone systems are being operated at once, you can use an antenna distribution system. An antenna distribution system reduces the total number of antennas needed and can help improve overall performance.

Antenna Distribution System

Rechargeable Battery Solutions

Unlike wired microphones, all wireless microphone transmitters require batteries. Some operate using standard AA batteries but many these days, like most Shure models, use a rechargeable battery solution to power the transmitters.

Shure's Lithium-Ion rechargeable batteries are robust, quality-tested, and engineered to ensure they stay healthy throughout their lifetime and use. Here are six key reasons to consider Shure rechargeable batteries:

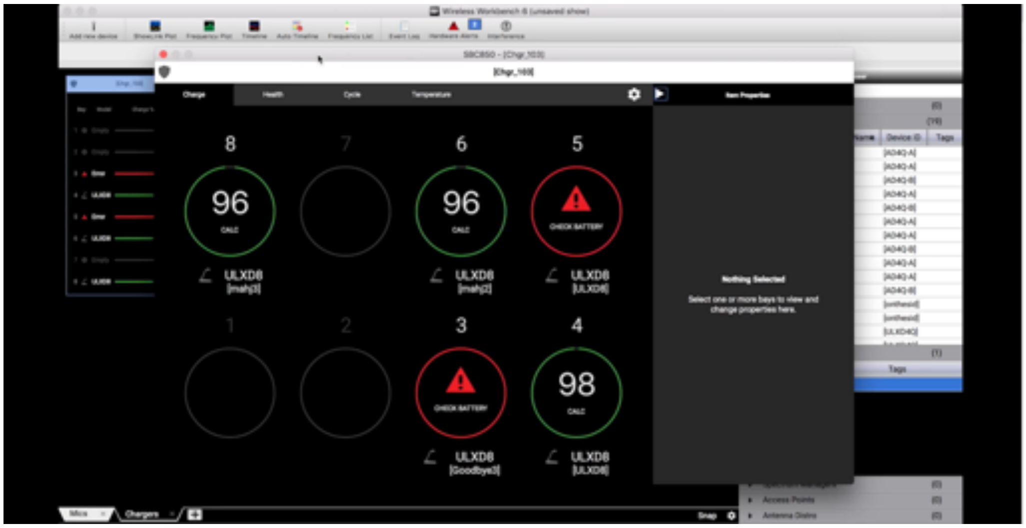
- Long-term investment that saves cost and pays for themselves over time
- Improved battery runtime through lithium-ion technology versus non-rechargeable batteries
- Smart battery technology provides insight into battery runtime, displayed in hours and minutes, and overall battery health statistics
- Network battery monitoring allows you to check battery status over the network for remote status updates
- Rapid charging and different charging options allow for workflow efficiency and keeping your products always at the ready
- Rechargeable batteries are more eco-friendly and decrease waste in the environment

Charging Options

Some stations charge lithium-ion batteries in or out of their transmitters.

Battery Workflow Tips

- Use Shure rechargeable batteries when possible. Shure batteries have no memory effect and provide a variety of recharging options to make the workflow easy and repeatable.
- Check your batteries before each service. Battery meters are provided on most transmitters and their linked receivers, as well as software for networked devices. For disposable batteries, many prefer using new batteries for each service to avoid any issues.
- For off-the-shelf batteries, Alkaline batteries are recommended since they provide longer, more consistent life than rechargeable or basic (carbon-zinc) batteries for wireless applications.



Monitoring Batteries

Track batteries in use or charging in a networked charger from Wireless Workbench software.

Longterm Rechargeable Battery Care

For storage longer than 3 months there are a few guidelines to consider. As a general rule, remove lithium-ion-powered devices and batteries when fully charged and store outside the charger for best lithium-ion battery health.

1. **Manage Charge Status** - As available for your rechargeable product batteries, use Storage Mode. When Shure rechargeable batteries will not be used for extended periods of time, Storage Mode will leave the battery in a slightly depleted state that is optimal for long-term storage. If your charger does not support Storage mode, then charge the batteries to full as you normally would. Do not store batteries that have been fully discharged.
2. **Store Batteries Separately** - Removable batteries should be taken out of any charger or device when storing for extended periods. If the battery is built into the device, then make sure any switches or power are set to off.
3. **Store at Room Temperature** - Batteries should be stored in a controlled, room temperature environment, so the temperature remains relatively steady. Normal HVAC with a thermostat is perfectly adequate. Large temperature fluctuations, or extreme hot or cold storage, will speed up and increase the aging of a lithium-ion battery.
4. **Long-term Storage** - If rechargeable batteries need to be stored for longer than one week, we highly recommend that they be recharged every six months using your Shure charger’s Storage Mode feature. If Storage Mode is not an option, recharging to full is an acceptable solution.

While these techniques can help maintain battery life, lithium-ion batteries do age and degrade over time. These guidelines are intended as a general explanation of how to handle Shure lithium-ion batteries. Always reference Shure product user guide and supporting documentation, and any applicable health and safety regulations, for any specific handling instructions.

Wireless Personal Monitor Systems

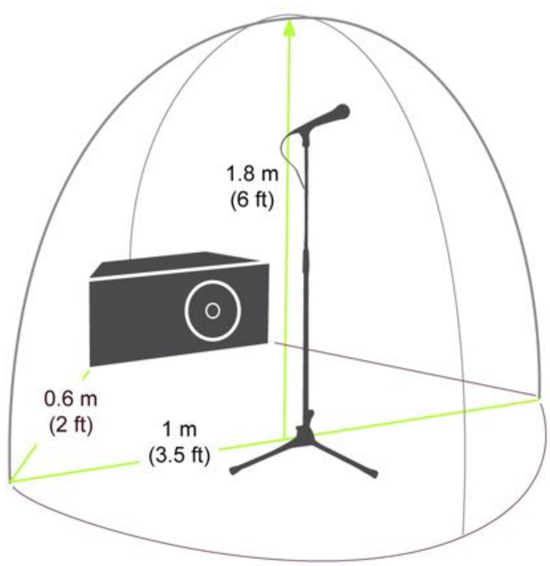
Introduction to In-Ear Monitoring

Praise and worship leaders and musicians all need to hear themselves as they speak, sing and play. Otherwise, they will have little idea if they are on key, on cue, or even on at all. For this reason, they need to monitor their sound. There are two main ways this is achieved: floor monitors or wireless personal monitoring systems. - and in-ear wireless monitoring systems, which has quickly become more commonplace with advancements in technology and more affordable options.

Floor Monitors - The Old Solution

Traditionally these monitors have been speakers (called *floor monitors or floor wedges*) aimed towards the individual – instead of towards the congregation – and often include just a portion of the overall mix.

As you have likely heard, the sound on the platform itself is usually loud, confusing, and requires the musician to stand in front of a monitor loudspeaker in order to make any sense of what he or she is hearing. The performer has to find the sweet spot in order to hear the right amount of desired program audio in relation to the surrounding ambient sounds.



Sweet spot created by a monitor wedge

However, there are some downsides to floor monitors that should be understood since they can be critical to the overall audio presentation:

- ***They are a major reason why the platform is so loud.*** So loud, in fact, that the main members of the worship team have trouble hearing and being heard. When musicians can't hear themselves and ask to have their monitor volumes increased, they frequently get involved in a “volume war,” creating an endless cycle of ever-increasing levels on the platform.

- ***The congregation in the first few rows can hear these speakers.*** This increases the overall volume of what they hear while decreasing the overall clarity – especially since they are hearing only parts of the full sound from in front of them and the entire sound from behind or sides. Since the congregation has to concentrate more to hear clearly, they get tired more quickly (this phenomenon is called listener fatigue) which dramatically decreases the overall impact of your service.
- ***They negatively affect the quality of the sound.*** Monitors can be reflected off a wall behind the platform and cause echoing and timing problems. Additionally they only provide a ‘mono’ sound to the people using them, making them inferior to other modes of monitoring.
- ***Floor monitors limit mobility,*** since the praise leader and the musicians must stand in a ‘sweet spot’ to hear themselves play.
- ***The monitors and cables used to operate them make for a messy platform,*** hinder line-of-sight for the people in the front, and add obstructions for the worship team.
- ***Floor monitors are the primary cause of feedback.*** The #1 reason for feedback is when a microphone picks up sound from a loudspeaker. Since the floor monitors point directly at people using microphones, the likelihood of feedback is considerable.
- They are heavy and hard to transport to other venues. This is a large concern for bands that take their worship on the road and for portable churches.
- ***There are hidden costs to floor monitors,*** since they also require amps and cables, as well as possibly an EQ system.
- ***Last, but far from least, floor monitors increase the risk of damage to your hearing.*** Most musicians like to turn their monitors up to hear themselves better, which, if done too much and too often, can lead to serious and permanent hearing loss.

Personal Monitoring - A New Standard

Now that personal monitoring systems are appearing on the platforms of even the smallest houses of worship, it’s a good time to understand their advantages, learn how to select the personal monitor systems for your needs, and finds ways to maximize your investment in this technology.

The Benefits to Personal Monitoring Systems

Personal monitors allow the pastor, praise leader, musicians and choir leader to personally hear just what they want without affecting what others hear. These systems are comfortable, wearable amplification devices that are designed to replace floor wedges with earphones that are worn 'in ear.'

The advantages for the people on the platform and the overall house of worship sound are numerous:

Greater control:

Personal monitors provide the ability to select precisely which mixes the user wants to hear. They allow the user to control the volume and balance of these mixes.

Some systems, like Shure PSM systems, let the users hear two different mixes and control the levels of these mixes.

Examples of this would include:

- The entire praise band as one mix AND the vocals as the second mix
- The sound from the platform as one mix AND the congregation through ambient microphones as the second mix
- The pastor (discreetly) as one mix AND the praise band as the other
- Or for the drummer: The praise band as one mix AND the click track as the other

Lower volume levels with higher sound quality:

When the musicians and others are 'in ear' they enjoy high-fidelity sound at lower volume levels and with less interruption from outside noise.

Let's say, for example, the music is especially loud (which could be good). Should a musician decide to increase their monitor volume ever so slightly, they can do so by using a control at their waist instead of having to signal to (and wait for) the sound engineer. Plus, the increase in sound cannot be heard by anyone else on the platform, which avoids any resulting 'volume wars' where the other musicians must now increase their monitors to hear over this additional platform noise.

A few words on the importance of hearing conservation:

We have touched on this earlier and we strongly believe that if you consider personal in-ear monitors for no other reason, you should consider them for hearing conservation benefits.

The potential danger of continual exposure to performance-level sound has been shown to cause permanent and significant hearing loss. This is why hearing conservation should be of serious concern to anyone who is a regular participant in auditorium-level sound and for those who recommend audio solutions.

Please note that people can still suffer hearing damage while using a personal monitoring system. Using any audio equipment improperly, without the limiter engaged, or at high-volume sound levels can be damaging to hearing.

For more information, talk to an audiologist or your doctor.

Decreased vocal strain:

In order to compete with the sound coming from the floor wedges, singers often sing louder than necessary. This causes vocal strain and, unless the sound engineer is able to lower this sound level, also decreases the quality of the music. When the singers have the ability to better adjust what they hear, they do not need to sing louder to hear themselves, so they can sing more naturally. This is better for their throats and, of course, for the congregation's ears.

You might notice that when some singers try personal monitors for the first time, they will have a tendency to "under sing." This is because they hear themselves so well now they believe they are singing loudly enough. A good trick here is to turn their mix down somewhat so that they will produce the necessary level from their voice.

Virtually no chance of feedback:

Feedback is caused by sound from loudspeakers leaking into live microphones. The louder the sound and the closer the speakers are to the microphones, the more likely you'll get degraded sound for the audience and, when the volume is too great, feedback.

Since personal monitors do not throw sound back towards the microphones, as wedges do, the chances for feedback from this source are eliminated.

Portability:

This is an important benefit, of course, for touring groups, but it is a major time and back saver for those churches which meet in rented spaces, such as schools, hotels, etc. Why lug around floor monitors, racks of amplifiers, equalizers (EQs), and cables, when you can have a small bag with your whole monitor system in it?

Greater mobility:

When the sound is directly in the musician's ear, it makes little difference where he stands on the platform … or off. He will hear the same mix at the same levels, which allows for more movement and interactivity. Suddenly, the entire platform is his “sweet spot”.

Obviously this is more of a benefit for those who choose wireless personal monitor systems.

Fewer platform perils:

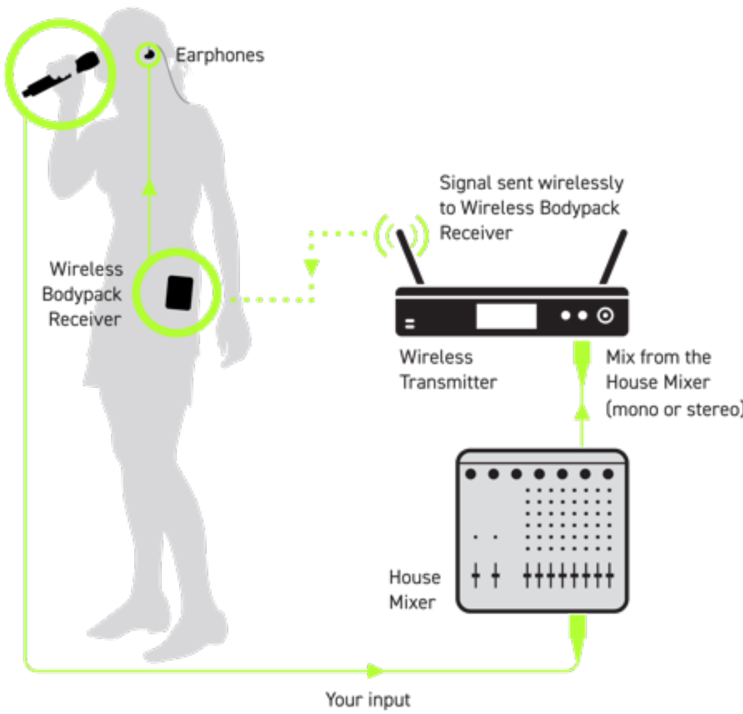
Here again, what you lose is what you gain. With personal monitors, you eliminate the floor wedges as well as the cables attached to them. This provides a cleaner, more aesthetically pleasing worship space with fewer boxes and cords to trip over.

The #1 reason to use Personal Monitors? Improved sound quality for everyone:

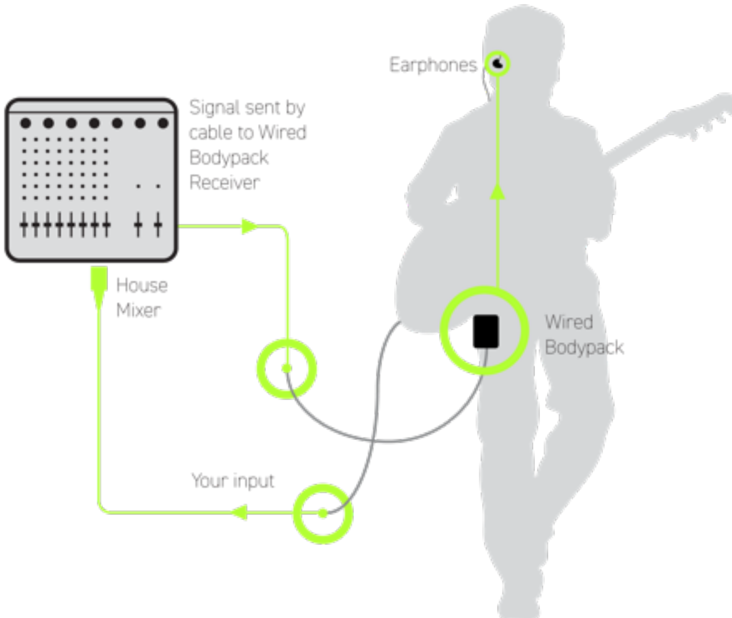
The most important benefit of using personal monitors in a house of worship is the overall improved sound clarity for everyone involved. All houses of worship, regardless of the acoustic challenges they often present, will benefit from personal, in-the-ear, monitor systems. And the benefits will be obvious to the congregation.

Example PSM Systems

Wireless Vocalist Setup



Wired Guitarist Setup



Earphones

There are a few reasons to consider using earphones in your house of worship, regardless of whether or not you decide to use personal monitoring systems:

1. They provide an improved level of sound clarity.
2. They are more aesthetically-pleasing than headphones.
3. They can easily integrate with other audio products you currently use.

Improved Sound Clarity

Like any sound system, any the premium components in a signal path are rendered ineffective by a low-quality listening device. You have already experienced this phenomenon with your smart phone or your computer speakers. The same holds true on the worship platform. When considering which listening device to use, you need to consider two key concepts: isolation and precise sound reproduction. In both cases, more is better.

Isolation

Whenever you see a singer put their hands over their ears – even when they are wearing headphones – you are seeing a symptom of a lack of isolation. Simply put, isolation is the ability of the listening device to eliminate background noise. Better isolation means fewer distractions from unwanted sounds and the ability to listen at lower – and safer – volume levels.

Vocalists will also tend not to ‘over sing.’ That is, they will not feel they need to compete vocally with what is coming into their ears. This, too, will result in more natural, textured vocals. For purposes of isolation, listening devices can be ranked in the following order, from best to worst:

Earphones– since these include sleeves that can precisely match the contours of the ear canal, they provide maximum isolation from background and ambient sound.

Headphones– the isolation provided by headphones varies considerably, depending on how well they cover the ears, their shape, the quality of the materials used, and the usage for which they were intended. (Note: many headphones were not designed for live sound or use in large, open spaces.)

Ear Buds– basically, these are tiny, often low-quality earphones that sit just inside the pinna of the ear. Ear buds are designed for aesthetic purposes or to meet smaller budgets, not primarily for sound quality. They usually provide very little isolation.

Another advantage to proper isolation: less “bleed through."

How often have you sat next to someone wearing headphones and you can hear their music almost as well as they can? This is called ‘bleed through’ and it is distracting on the platform and in the congregation as well. Proper isolation lets the listener hear at lower volumes, decreases the overall volume required, and isolates the sound from others.

Earphones are More Aesthetically Pleasing

As house of worship sound becomes richer and more complicated, the people on the platform need to isolate their own sounds better. This has resulted in more and more people, especially the lead and backup singers, wearing headphones. While helping the singers provide richer sound, the headphones are distracting to the congregation and make it harder for the worshippers to connect, personally, with these members of the worship team.

More connection for the congregation

Earphones fit snugly in the ear and utilize thin cables that go under the collar so they cannot be seen at all from the congregation. And, as discussed above, they provide superior isolation, so the vocalists can sing at more natural levels.

So, Who Gets Earphones?

All personal monitor system users should include earphones as part of their system, so the praise band, the choir leader and officiant should already be ‘in ear.’ If not, and they wear headphones, you should replace these headphones with earphones as they look more natural, receive better quality audio at lower volumes, and do not distract others with their ‘bleed through,’

While the same argument can be made for giving earphones to everyone on the platform who now wears headphones, it really comes down to your budget.

Unless you are purchasing high-end headphones, you will probably spend more per set on earphones than you do on headphones.

Earphones Can Be Used with Other Audio Products

Earphones can replace headphones in nearly all applications, such as:

- **Personal monitor systems that did not come with isolating earphones**. There is no reason you have to use the headphone or earbud supplied with the system you have now. You can add all the advantages of isolating earphones by simply unplugging the current headphones and plugging in the earphones you want.
- **Assistive listening systems**. Here, again, simply use isolating earphones instead of the headphones provided.

While you might need to have a container of sleeves on hand for congregation members using assistive listening systems, you should find that the lower-profile and increased sound quality (for both the user and the people sitting near the user) are well worth the added effort.

- **Consumer products**, including smartphones, computers, laptops, and music players. Isolating earphones have become extremely popular for people who enjoy hearing the subtleties of their music. This means you can enjoy better sound quality when you are away from the house of worship and get more value from every set of earphones you purchase.

Earphone Foams and Sleeves

While the earphone is a critical component to any personal monitor system, there is a component to the earphone that is just as critical to the entire in-ear monitoring experience: the sleeves.

Made from rubber or foam, these ‘sleeves’ attach to the end of the earphone and are the only part of the system that makes direct contact with your ear. For this reason, they must be comfortable, secure, and isolate correctly.

Some personal monitor systems come with a collection of these sleeves in various sizes. Since everyone’s ears are different, finding the proper sleeves is, possibly, the most important aspect to getting the best sound from your personal monitor system.

Replaceable Ear Sleeves

Many earphones come with a collection of sleeves in various styles and sizes.

It is important to consider all of the following tips and techniques:

- The earphones should come with a number of sleeve options such as foam and rubber, as well as small, medium, and large. Make sure all the people using earphones try all the various sizes and types, not just the ones that “look” right.
- Of the sleeves that come with the earphones, the foam type that expand to fit the ear canal usually provide the most isolation. Make sure everyone tries these before settling for rubber ones.
- Consider a custom-molded sleeve. Talk to your audiologist or contact a company that provides these. Since they will be made to precisely fit the user’s ear canals, they will provide the best combination of isolation and comfort.

Questions About Earphones

How do I know I have the earphones in correctly?

First… are they comfortable? They should be snug, but not painful to wear. Second… do they provide the isolation you require? You should be able to listen to your mix at fairly low levels without distraction from the other sounds on the platform.

Should I use only one earphone or should I keep both in while I sing?

For optimum performance and hearing protection, you really should wear both earphones. Removing one of the earphones will take away many of the system’s advantages. If you feel detached from the worship, there are ways to mix in ambient mics to eliminate this isolation.

Everything sounds ‘hollow’ to me but no one else is having this problem. Is it me?

It’s not you, but it might be your ears. Everyone’s ears are different and everyone hears sound somewhat differently. Try different sleeves until you find the best fit. Using foam sleeves (instead of rubber ones) is usually a good way to solve this, but you might also want to look into having custom sleeves made to fit. Companies such as Sensaphonics can provide custom sleeves and help answer any questions you may have.

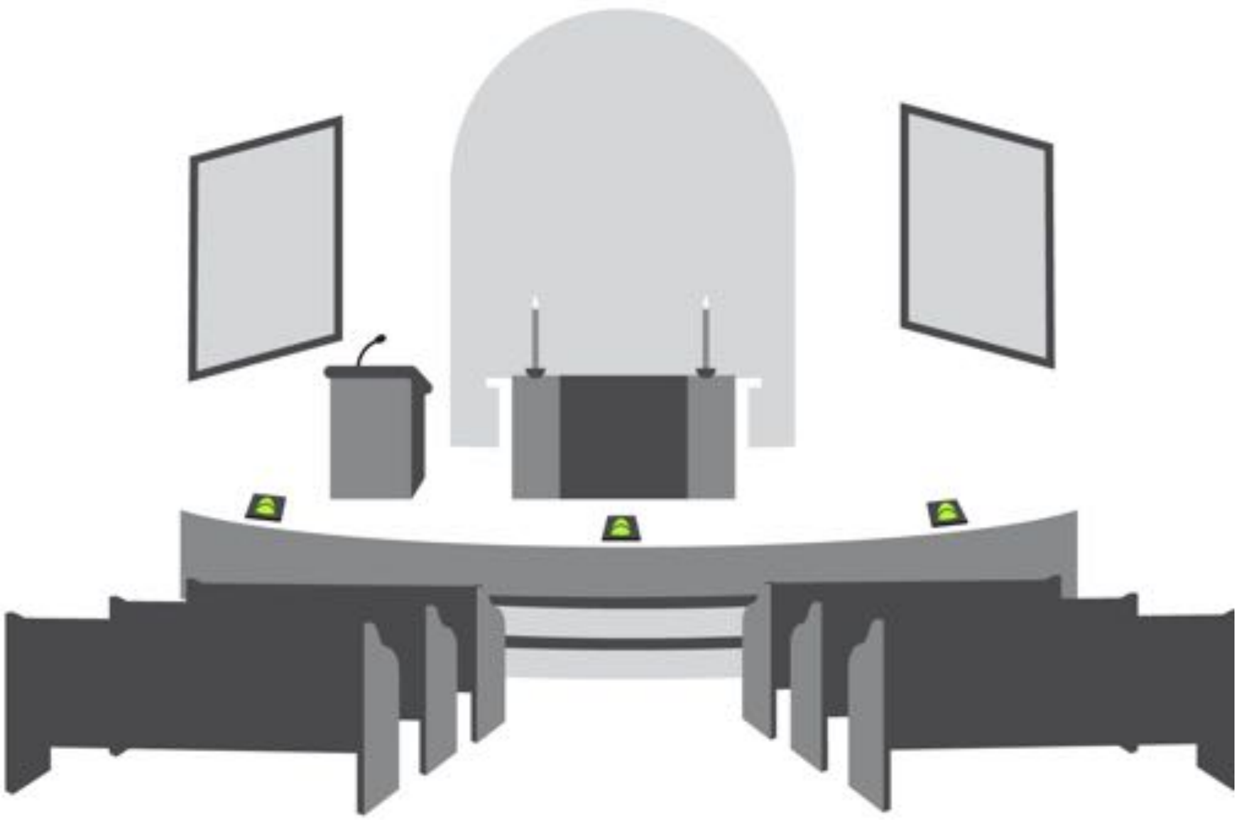
Optional Components

While the praise and worship team members can get all the advantages of using personal monitors ‘straight out of the box,’ there are a few components that are worth considering.

Ambient microphones and other ways to solve isolation issues

Personal monitors are designed to help improve the overall sound and provide isolation from platform noise and other distractions. It is this ability to create isolation from outside sound that allows people to listen to their mix at a more comfortable level. This level is typically lower than that of the platform.

One of the challenges, however, is to make sure the people on the platform are not entirely cut off from the service. This is achieved by setting up ambient microphones to capture the sound and feel of the room and congregation (sounds participants naturally hear when they aren't wearing in-ear monitors).



Example mic placement for ambient and congregation sounds

Because personal monitoring using in-ears provides so much isolation, adding some ambient audio provides a more natural mix for the listener. In this example, boundary microphones are spread across the stage and facing the audience.

Wired and Wireless Options

These receivers are capable of working with either wired or wireless transmitters. This lets you match your monitor configuration to the particular need of the musician, since there are times when a wired system is the better choice (e.g. a drummer). Or, when RF spectrum is crowded in your environment, the wired cable frees up valuable frequencies. Additionally, you can combine use of wired and wireless at the same time for added flexibility and adaptability (such as a click track plus a mix of the service).

Hardwire Version for PSM900

Personal Monitor Mixer

This type of mixer gives the user control over his or her own mix without affecting the signal path to the main house mixer.

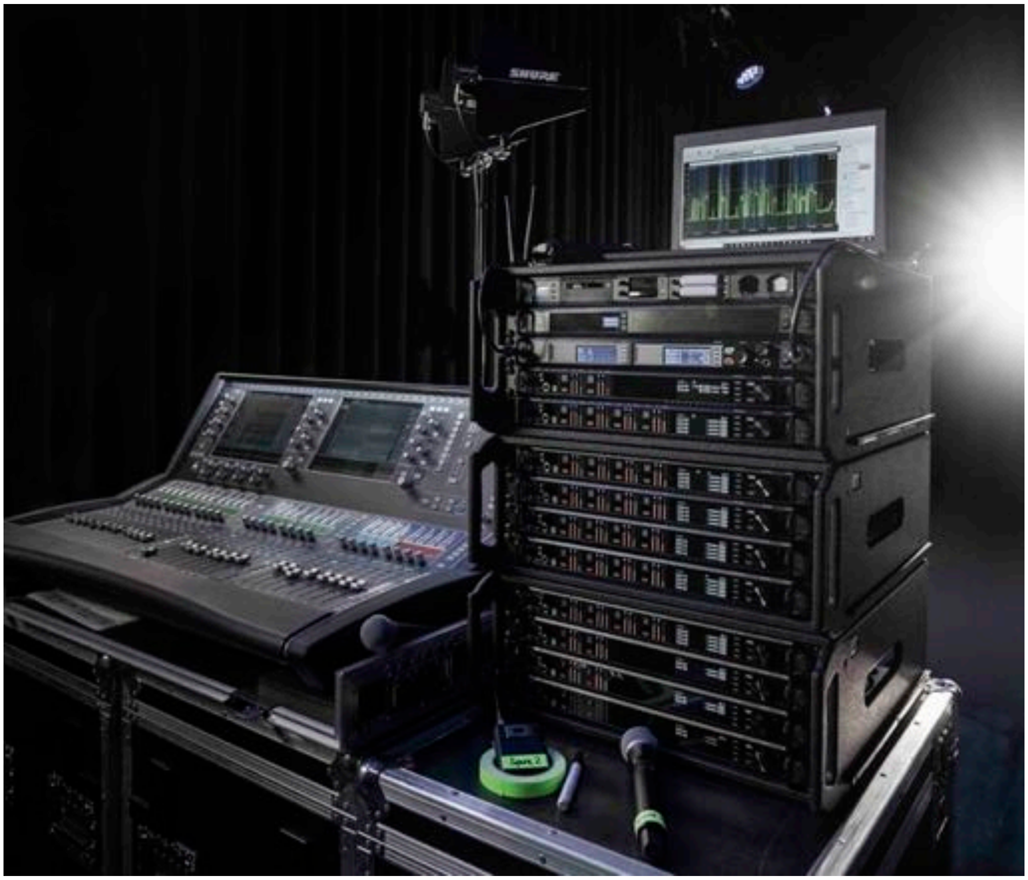
A personal monitor mixer puts more control on the platform instead of relying so heavily on the person at the soundboard … if that person even exists. Also, once set, the personal monitor mixer 'remembers' these settings so your praise ministry can have the same mix at every service.

This is of particular interest to 'portable churches' (churches that rent space on an hourly basis), which require fast set-up and tear down, but refuse to sacrifice sound quality for speed.

A personal monitor mixer is also useful for praise bands that travel and want to bring their pre-set mixes with them. This allows the control they need without relying too heavily on the person running the sound at the house of worship.

Digital mixer

When used with personal monitors, digital mixers can recall your individual mixes time after time, exactly as they were re-hearsed in every environment regardless of any acoustic challenges.



Example of a digital mixer

Additional Applications for Personal Monitor Systems

The following sections provide a few additional uses that are applicable to houses of worship. We expect that once you have decided to include personal monitors in your house of worship, you will find many more.

Enable discreet communications

Pastors, praise leaders, and choir leaders can receive spoken cues and information during the service, either from others on the platform or even people elsewhere in the house.

Here are two examples of how this might work:

1. Provide your pastor with a lavaliere microphone that goes only into the praise leader’s mix. In this way, the pastor can give the praise leader cues, such as changing the hymn or increasing the length of a particular song.
2. Add the producer’s microphone to the pastor’s mix. Now the pastor can receive cues from the person responsible for making sure the service runs smoothly.

Other times a user might want to receive spoken cues include:

- Providing timely information or details to the pastor or others during community meetings.
- To prompt on-stage directions or missing lines to actors in theatre productions and skits.
- Whenever you feel that someone might need to receive information discreetly.

Distributed Audio

Distributed audio refers to devices that accept analog audio from a mixer or other source, convert this signal to digital audio, then send this signal to a destination via Category 5 (CAT-5) cable (which is high performance digital transmission cable also used for Ethernet connections). At the end point it can be converted back to analog and used however necessary.

The devices at the receiving end of the distributed audio network can be DSP devices, digital mixers, or an individual multi-channel "personal mixer" for monitoring. The advantage of using a distributed audio system is the increased distance allowed between devices, plus the fact that CAT-5 lines are often already installed in many facilities.

Distributed audio therefore becomes very useful for monitoring systems since getting multi-channel mixes to many people becomes that much easier. Also, you can transmit the personal mix to anyone who wants the added mobility of wireless but also wants a customized multi-channel mix they can control with the personal mixer.

Cue to the next part of the service.

More and more houses of worship are including pre-recorded music or events into their services. Personal monitor systems allow the wearers to hear these recorded events as they are being faded into the service. The musicians can soften their music accordingly or add any lead-ins right on cue. This provides the congregation with a more seamless experience.

Bring the service to a conference room or the nursery

Personal Monitor Systems can also be used to bring audio to another part of your church. Let’s say you want to bring the sound of the service to the Nursery and you don’t want to punch holes through the walls to lay speaker wire…

It’s just like setting up a wireless mix for a musician on the platform. Place the bodypack for the personal monitor system onto a powered loudspeaker in the nursery. Insert the personal monitor’s earphone jack output into the loudspeaker’s input. This should let the people in the nursery hear the entire mix.

If you add ambient sound [See “Add the congregation to the mix” later in this chapter] you can provide the Nursery – or any room – with the complete audio experience.

Not only do you have halls without walls, you have just created what is technically referred to as a "Point-to-Point" wireless system.

By using a ‘battery eliminator,’ you can power the bodypack from an electrical outlet, which will save you from having to ever replace batteries.

Better rehearsals

It is very infrequent that the praise band has the luxury of rehearsing on the actual platform. Personal monitors can quickly turn the worst rooms into a great place to practice, allowing you to hear more clearly by virtually eliminating the rehearsal room’s poor acoustics.

Add the congregation to the mix

With a few strategically placed microphones, you can add the sounds of the congregation to the mix that is being sent to the personal monitor systems.

Once you have done so, the praise leader and other musicians will be able to hear the congregation without having to resort to removing one of the two earphones, which should be discouraged since doing so eliminates most of the benefits that the personal monitors provide.

Some tips and techniques for ambient miking:

- Place the ambient microphones on the edges of the platform facing the house. It is best to position the microphones in front of, above, and aimed towards the faces of the congregation.
- Do not place the microphones in the congregation.

- Make sure the microphones are properly oriented, so the microphones send signals to the correct ear. For example: It is important that sounds from the left side of the house are heard in the left ear of anyone monitoring the sound.
- When selecting which microphones to use, treat the congregation as you would a large group of singers. Condenser microphones with a cardioid polar pattern are usually best.
- Do not be tempted to use shotgun microphones.
- Overhead (ceiling-mounted) microphones can be used, but are often far less effective and harder to control than on-platform microphones.

Let the musicians 'feel' the music

Try a ‘buttkicker’ (also called drum throne shaker) to recreate the vibrations that drummers and bass players hear and feel when low-frequency sounds are amplified. Placed on the user’s stool or beneath a riser, they provide physical vibrations along with the music.

Getting the Best Sound

Microphone Techniques for Houses of Worship

In order to select a microphone for a specific application, 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 previously discussed will lead to an appropriate match. Finally, correct placement and proper use will insure best performance. In this section, we will present recommendations for some of the most common worship facility sound applications. The sound system in the following examples is assumed to be of high quality, with balanced low-impedance microphone inputs and available phantom power.

Lectern

The desired sound source for a lectern microphone is typically a speaking voice, though one may occasionally be used for singing. Undesired sound sources that may be present are nearby loudspeakers (possibly a central cluster overhead), and ambient sound (possibly ventilation or traffic noise, and reflected sound).

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 better satisfied by a condenser design, which can maintain high performance even in very small sizes. Dynamic types are somewhat larger, but they do not require phantom power.

To match the desired sound source (the voice), the microphone must have a frequency response that covers the vocal range (approximately 100Hz to 15kHz). Within that range the response can be flat, if the sound system and the room acoustics are very good; but often a shaped response, with some presence rise, will improve intelligibility. Above 15kHz and below 100 Hz, the response should roll off smoothly, to avoid pickup of noise and other sounds outside of the vocal range, and to control 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 likelihood of feedback since it can be aimed toward the talker and away from loudspeakers. Depending on how much the person speaking may move 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 slightly 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. Sufficient sensitivity for lectern use can be achieved by either condenser or full-size dynamic types, since the sound source is fairly strong and picked up from only a slight distance.

The physical design of 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 8 to 16 inches away from the mouth, and aimed toward the mouth. This will guarantee good pickup of the voice and maximum rejection of unwanted sources. Locate the microphone a few inches off-center and below the mouth level. This will greatly reduce breath noise that occurs directly in front of the mouth but will still provide good coverage throughout the pickup angle of the microphone.

If possible, adjust the sound system to provide stable operation with the lectern microphone at a nominal distance of 12 inches. This will provide relatively less change in level with changes in distance than if the microphone is placed much closer, due to the inverse square law. For example, with a nominal distance of 12 inches a change of ±6 inches results in a -3.5dB to +6dB level change. For a nominal distance of only 6 inches, the same distance change results in a -6dB to greater than +18dB level change, a much larger variation. The difference in potential acoustic gain between the two nominal positions is 6dB.

For proper operation, the microphone must be connected to the sound system with quality cables and connectors. The correct phantom power should be applied if a condenser microphone 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 and acoustic noise. Goosenecks should be quiet when flexed. It is strongly recommended that a pop filter be placed on the micro phone to control explosive breath sounds, especially when using miniature condenser types.

Good techniques for lectern microphone usage include:

- Do adjust the microphone position for proper placement.
- Do maintain a fairly constant distance (8-16 inches).
- Don’t blow on microphone, or touch microphone or mount, in use.
- Don’t make excess noise with materials on lectern.
- Do speak in a clear and well-modulated voice.

Altar

Boundary microphones are useful for the altars, lecterns, tables, and other flat surfaces. The wireless version as shown reduces clutter and the need to make holes in costly or historical furniture.

The desired sound source for an altar application is a speaking (or sometimes singing) voice. Undesired sounds may include direct sounds, such as choir, organ, or loudspeakers and ambient noise sources, such as building noise or the congregation itself.

A boundary microphone is the physical design best suited to this application. Its use will minimize interference effects due to reflections from the altar 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 optimized for the vocal range and will benefit from a slight presence rise. A unidirectional (typically cardioid) pattern will give the broadest coverage with good rejection of feedback and noise. A condenser microphone will provide the highest sensitivity. Finally, the microphone should have a balanced low-impedance output.

Good techniques for altar microphone usage include:

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

The microphone should be placed flat on the altar at a distance of 2 to 3 feet and aimed towards the normal position of the person speaking. It should be located or aimed away from other objects and from any local noise such as page turning. Unless there is more than one distinct position to be covered, and unless these positions do not violate the 3-to-1 rule, use only one altar microphone.

The microphone should be connected and powered (if a condenser) in the proper fashion. If the altar 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. Do not cover the microphone with heavy altar linens.

Handheld Vocal

The desired sound source for a handheld microphone is a singing or speaking voice. Undesired sounds may include other singers, musical instruments, and various ambient sounds. In addition to the normal loudspeakers, the sound system may also have nearby “monitor” speakers aimed toward the singer.

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 available for high-performance applications. The preferred frequency response is shaped: vocal range, with presence rise for intelligibility and low-frequency roll-off for control of proximity effect and handling noise. These microphones should always be 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 adequate sensitivity may be achieved with dynamic or condenser types. Finally, the physical design is optimized for comfortable handheld use, and generally includes an integral windscreen/pop filter and an internal shock mount. An on-off switch may be desirable in some situations.

Handheld Vocal Application

Positioning a handheld microphone at a distance of 4 to 12 inches from the mouth (and aimed towards it) will give good pickup of the voice. In addition, locating the microphone slightly off-center, but angled inward, will reduce breath noise.

With high levels of sound from adjacent musical instruments or other singers, it may be necessary to hold the microphone closer to the mouth. If the distance is very short, especially less than 4 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 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 handheld microphones. A stand or holder should also be provided if it is desirable to use the microphone hands-free.

Good techniques for handheld microphone usage include:

- Do hold microphone at proper distance for balanced sound.
- Do aim microphone toward mouth and away from other sound sources.
- 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 dynamics with voice rather than moving microphone.

Lavalier

The desired sound source for a lavalier microphone is a speaking (or occasionally singing) voice. Undesired sources include other speaking voices, clothing or movement noise, ambient sound, and loudspeakers.

Lavalier Application

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. In particular, the consonants “d”, “t”, and “k” create strong downward breath blasts that can result in severe “popping” of unidirectional lavalier microphones. Placing the microphone slightly off to the side (but still aimed up at the mouth) can greatly reduce this effect.

Balanced low-impedance output is preferred as usual. Adequate sensitivity can be achieved by both dynamic and condenser types, due to the relatively close placement of the microphone. However, a condenser is generally preferred. 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 allow mobility for the user.

Placement of lavalier microphones should be as close to the mouth as is practical, usually just below the neckline on a lapel, a tie, or a lanyard, or at the neckline in the case of robes or other vestments. 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 connectors, and provide phantom power if required.

Good techniques for lavalier microphone usage include:

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

Headworn

Again, the desired sound source for a headworn microphone is a speaking or singing voice. Undesired sources include other voices, instruments, ambient sound and sound system loudspeakers.

Most headworn microphones are of the condenser type because of their small size and superior sound quality. A dynamic type can be used for speech-only applications or if larger size is not an issue. For either type, the frequency response is shaped for closeup vocal with some presence rise. An omnidirectional polar pattern is suitable for most applications, especially if the microphone does not reach all the way in front of the mouth. A unidirectional pickup is preferred in very high ambient noise applications or to control feedback from high volume monitor speakers. For proper operation, unidirectional types should be positioned in front of or directly at the side of the mouth and aimed at the mouth. A windscreen is a necessity for a unidirectional headworn microphone.

Balanced low-impedance output is preferred for hardwired setups but headworn types are often used in wireless applications. In that case, the impedance and wiring are made suitable for the wireless system. For condenser types, the bodypack transmitter provides the necessary bias voltage for the microphone element.

There are many different headworn mounting designs. Most have a headband or wireframe that goes behind the head, while a few are small enough that they merely clip over the ear. In all cases, the microphone element is at the end of a miniature “boom” or flexible arm that allows positioning close to the mouth. Again, an omnidirectional element can be positioned slightly behind or at the side of the mouth while the unidirectional type should be at the side or in front and aimed toward the mouth.

The main advantages of the headworn microphone over the lavalier are greatly improved gain before feedback and a more consistent sound level. The increase in gain before feedback can be as much as 15- 20 dB. This is completely due to the much shorter microphone-to-mouth distance compared to lavalier placement. The headworn can nearly rival a handheld type in this regard. In addition, the sound level is more consistent than with the lavalier because the headworn microphone is always at the same distance to the mouth no matter which way the user may turn his head.

Headworn Application

Good techniques for headworn microphone usage include:

- Do observe proper placement and orientation.
- Do adjust for secure and comfortable fit.
- Don’t allow microphone element to touch face.
- Do use pop filter as needed, especially for unidirectional.
- Do adjust vocal “dynamics” to compensate for fixed mouth-to-microphone distance.

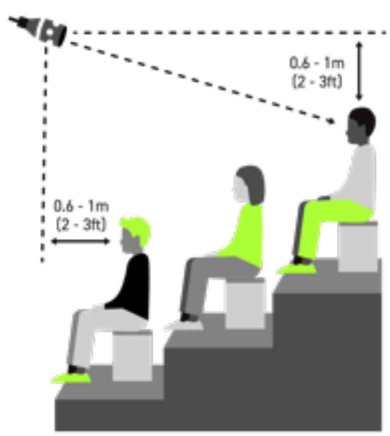
Choir

The desired sound source is a group of singing voices. Undesired sound sources may include the organ or other musical instruments, loudspeakers, and various ambient noise.

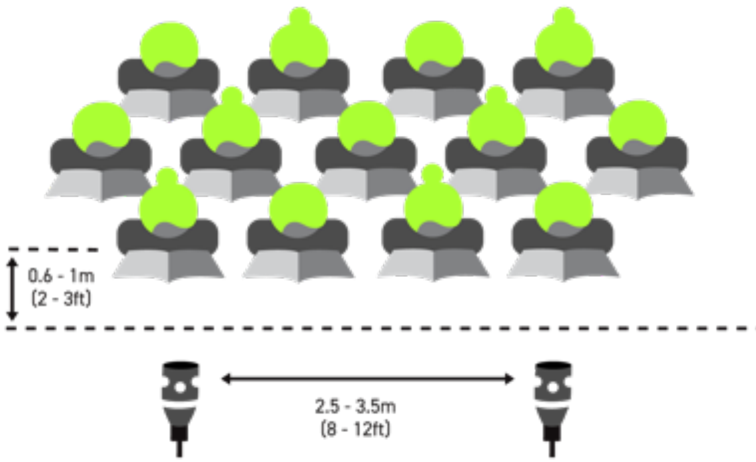
A condenser is the type of microphone most often used for choir application. They are generally more capable of flat, wide-range frequency response. The most appropriate directional type is a unidirectional, usually a cardioid. A supercardioid or a hypercardioid microphone may be used for slightly greater reach or for more ambient sound rejection. Balanced low-impedance output is used exclusively, and the sensitivity of a condenser microphone is desirable because of the greater distance between the sound source and the microphone.

The physical design of the microphone for choir pickup should lend itself to some form of overhead mounting. It may be supported by its own cable or by some other fixture, such as a stereo microphone mount. Finally, it may be a full-size microphone or a miniature type for unobtrusive placement. Application of choir 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 (or a large sound source) with one (or more) microphone(s). Obviously, this introduces the possibility of interference effects unless certain basic principles (such as the “3-to-1 rule”) are followed, as discussed to the right.

For one microphone picking up a typical choir, 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 choir and aimed at the last row. In this configuration, a cardioid microphone can cover up to 15-20 voices, arranged in a rectangular or wedge-shaped section.



Microphone Positions - Side View



Choir Microphone Positions - Top View

For larger or unusually shaped choirs, 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 choir 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 choir pickup, remember the following rules: **observe the 3-to-1 rule; avoid picking up the same sound source with more than one microphone; and finally, use the minimum number of microphones**.

For multiple microphones, the objective is to divide the choir into sections that can each be covered by a single microphone. If the choir has any existing physical divisions (aisles or boxes), use these to define basic sections. If the choir is grouped according to vocal range (soprano, alto, tenor, bass), these may serve as sections.

If the choir is a single, large entity, and it be comes 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 12 feet. If the choir is unusually deep (more than 5 or 6 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**.

It is very important to locate choir microphones as far away from loudspeakers as possible. Be aware of the rear pickup of supercardioid and hypercardioid types when aiming microphones. Try to avoid pickup of organ pipes or speakers in the choir loft. And, of course, keep microphones away from other noise sources such as air ducts.

Once overhead microphones are positioned, and the cables have been allowed to stretch out, they should be secured, if necessary, 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 the highest-quality cables and connectors, particularly if miniature types are specified.

Good techniques for choir microphone usage include:

- Do place the microphones properly.
- Do use the minimum number of microphones.
- Do turn down unused microphones.
- Do let the choir naturally “mix” itself.
- Don’t “over-amplify” the choir.
- Don’t sing “at” the microphone.
- Do sing in a natural voice.

The use of choir microphones is governed, to some extent, by the intended destination of the sound. In general, high-level sound reinforcement of a choir within the main body of the worship facility is not recommended. In fact, it is not possible in most cases, unless the choir itself is isolated from the main body of the worship facility. Use of area pickup microphones in the same acoustic space as area coverage loudspeakers results in severe limitations on gain-before-feedback. The best that can be done in this circumstance is low-level reinforcement in the immediate area, and, possibly, medium-level reinforcement to distant areas, such as under balconies or in 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.

Many older worship facilities are very reverberant spaces, which provide natural, acoustic reinforcement for the choir, though sometimes at the expense of speech intelligibility. Some modern architecture has been designed to provide a less reverberant space, both for greater speech intelligibility and to accommodate modern forms of music. This results in a greater reliance on electronic reinforcement. However, it is still not practical (and probably not aesthetically advisable) to make a choir of 20 sound like a choir of 200. The sound system (and the microphones) can provide some useful enhancement, but a large acoustically dead worship facility simply requires a large live choir.

Congregation

The desired sound source for a congregation microphone is a group of speaking or singing voices. Undesired sources are usually the sound system loudspeakers and various ambient sounds.

Condensers are the choice for highest-quality sound at a distance. A flat, vocal-range frequency response is usually desirable, with a unidirectional polar pattern to minimize pickup of unwanted sound. The electrical output should be balanced low-impedance, and the physical design should accommodate overhead mounting, by cable or other fixture. The microphone may be either full-size or miniature, depending on visual requirements.

Since this application of microphones is another example of area coverage, the placement should be in front of, above, and aimed toward the faces of the congregation. Though similar in concept to the choir example, fewer and more distant microphones may be used to pick up the overall ambience of the congregation.

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 can only respond to the sound in its immediate vicinity. If this local soundfield is louder than the distant sound from below, there is no hope of picking up a usable sound with a ceiling-mounted microphone.

Congregation area microphones are used exclusively for recording, broadcast, and other isolated destinations. It is never intended to be mixed into the sound system for local reinforcement. If it is desired to reinforce an individual member of the congregation, it can only be done successfully with an individual microphone in the congregation: a stand-mounted type that the member can approach or a handheld type (wired or wireless) that can be passed to the member.

Good techniques for congregation microphones include:

- All the techniques for “Choir Microphone Usage.”
- Do use only at a level sufficient to add ambience to a recording.
- Don’t mix area microphones into the sound reinforcement system.

Musical Instruments

A tremendous variety of musical instruments is used in current worship facility services. In fact, almost any instrument that exists may be used: from classical symphonic instruments, to modern electronic instruments, to historical and ethnic instruments of any description. Presented here will be techniques for three musical instruments that are widely used today: the acoustic guitar, the piano, and the organ.

Use of microphones with many other instruments is discussed in Shure’s Guides to Microphone Techniques.

Match the Microphone to the Sound Source

In each of these examples, the desired sound source is the musical instrument itself. Possible undesired sound sources include other nearby instruments, singers, loudspeakers, and the usual ambient noise sources.

Since accurate, wide-range reproduction of musical instruments is the goal, the use of condenser microphones is often preferred, although certain instruments, such as drums, can be well suited to dynamic types. The frequency response is usually flat and wide-range, especially for organ or piano. Unidirectional designs are preferable, to minimize pickup of undesired sound. Again, balanced low-impedance models are the best choice. Because close microphone placement is used, dynamic and condenser types have suitable sensitivity for general sound reinforcement. However, condensers are recommended for highest-quality sound. The physical design, though, can vary widely in instrument applications, depending on the desired placement and use.

Good techniques for an acoustic instrument microphone usage include:

- Do experiment with placement for best sound.
- Do maintain a constant distance.
- Do use a shock mount if stage noise is present.
- Don’t position microphone where it may be struck by instrument.
- Don’t allow voice to be picked up by instrument microphone.

Acoustic Guitar

Guitar Application

The acoustic guitar is a relatively small sound source that can normally be picked up quite well by only one microphone. Since most of the sound comes from the sound hole and the top of the guitar, a microphone positioned in front of the guitar can get an excellent overall sound. This sound will vary, however, as a function of the microphone distance from the sound hole and its orientation to the top of the guitar. The sound will be louder and “bassier” the closer to the sound hole; softer and thinner farther away. Proximity effect will also increase the bass response at close distances.

A full-size microphone can be positioned on a stand to give the desired sound. An alternate approach is to mount a miniature microphone directly on (or in) the guitar by means of a clip or holder. This keeps the microphone at a constant distance, and allows freedom of movement for the performer, especially if used with a wireless transmitter. In either case, care must be taken to position the microphone to avoid interfering with the player.

Piano

Piano Application

The piano is a relatively large acoustic source whose sound comes from the soundboard, the strings, and reflections from the lid and other body parts. Although the piano is normally heard at a distance, it is not feasible to use a distant microphone on a piano for sound reinforcement, due to gain-before-feedback limitations. Placing the microphone close to or inside of the piano is the normal procedure. The resulting sound is not entirely natural, but careful microphone placement can yield very good results.

Depending on placement, several microphone physical designs may be used. A conventional, full-size microphone can be positioned close to or inside of the piano (with the lid open) using a stand and boom. A position over the treble strings will yield a bright sound while a position over the middle or low strings will correspond to a bassier sound. A sharper attack is heard near the hammers, while a softer sound is heard farther away. For greater isolation from other sounds and to reduce feedback, a boundary microphone is sometimes attached to the underside of the lid, which is then partially or completely closed.

Since very close microphone placement may not pick up the full sound of a large source, it is sometimes desirable to use two (or more) microphones, especially for stereo reproduction. In this case, microphone placement becomes more subjective due to the possibility of interference effects. A good starting point is one microphone over the treble strings and a second over the bass strings. This will often produce a more balanced sound, and does allow a greater range of control. However, some experimentation will be necessary to get the best sound from a specific instrument in a specific room.

Good techniques for piano microphone usage include:

- Do experiment with placement for best sound.
- Do adjust lid for best sound and/or isolation.
- Do use shock mounts if vibration is a problem.
- Do listen for interference effects with multiple microphones.
- Don’t allow voice to be picked up by instrument microphone.

Organ

The organ is potentially the largest sound source in some worship facility applications. However, pipe organs and large electronic organs are not normally reinforced by sound systems, but rather are picked up for recording or broadcast purposes. Since the organ is also the widest range instrument, the careful placement of high-quality microphones is essential for best results.

A large organ produces sound from many ranks of pipes, or, for an electronic type, from a number of tone cabinets. Since it is not possible to use microphones on individual pipes or loudspeakers, some type of area coverage must be employed. Often, the groups of pipes or tone cabinets are widely separated, sometimes even located on opposite sides of the worship facility, as is the case with antiphonal ranks. This will require a decision on the goal of the sound.

If the goal is to reproduce the sound as heard by a listener in the house of worship, one or two (for stereo) microphones can be positioned in the body of the worship facility, over the congregation, and aimed toward the main organ ranks. This will pick up a representative organ sound, with a high proportion of room sound (ambient sound), as well as the sound from the choir and from the sound system itself. If the room has reasonably good acoustics, and if the level of the organ is well-balanced with both the choir and the sound system, this is the simplest and most effective way to simulate being in the religious facility. In some arrangements, the choir microphones themselves will pick up a suitable organ sound.

On the other hand, if the goal is to reproduce a concert organ performance that does not rely heavily on the room acoustics, or if it is desired to control the level of the organ independently of the choir and other sounds, it is necessary to place microphones to pick up the organ sound only. This will require that a microphone be placed close enough to each of the main locations of pipes or tone cabinets so that the microphone hears primarily the local organ sound, rather than the ambient or room sound.

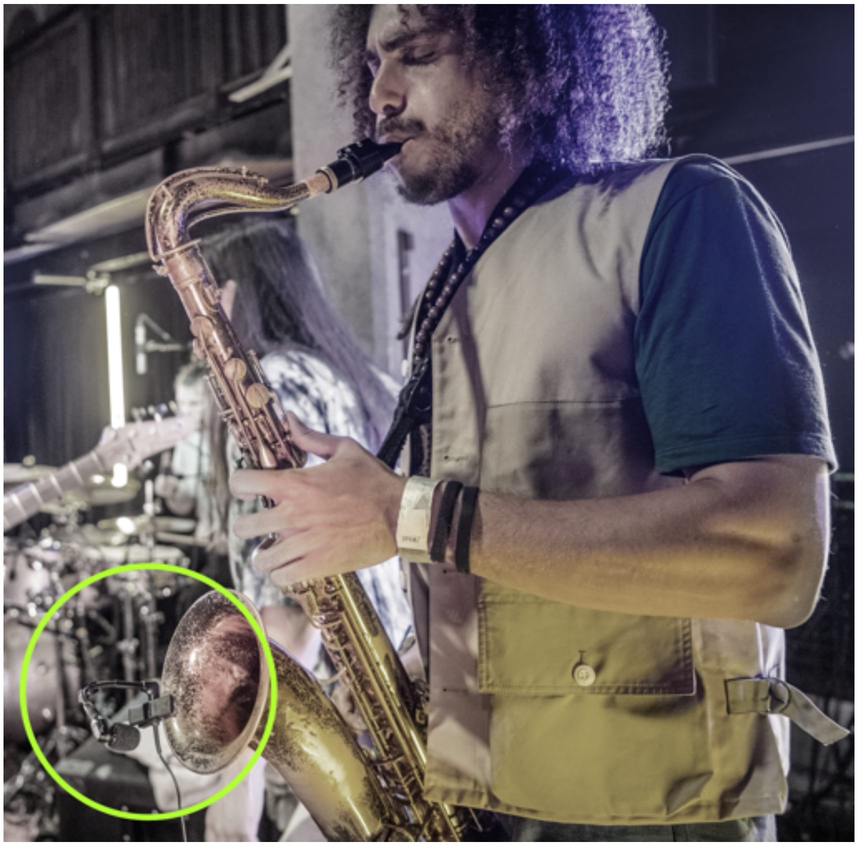
This may involve several microphones, depending on the number and location of sound sources. Individual placement should be done according to the guidelines given earlier with respect to choir pickup, although it may be possible to mount microphones on stands in organ galleries as well as overhead in front of exposed ranks. In any case, some experimentation with microphone positioning, and careful mixing of microphone signals will be necessary to get a full, balanced sound.

Horns or Woodwinds

The sound from brass instruments is very directional. Placing the mic off-axis with the instrument's bell will result in less pickup of high frequencies – leading to the sound being more diffused, with fewer upper harmonics and not as much bite.

By placing a mic on a brass instrument, on-axis – pointed straight at the bell – you can produce a bright, sharp, and clear sound.

By close-miking, with a clip on microphone for example, the sounds will be "tight," and you can minimize feedback and leakage. By placing the mic more distant, you will get a fuller, more dramatic sound.



Clip-On Microphone

Clip-on instrument mic and a bodypack transmitter allows movement and mobility

Other Musicians or Participants

Since not everyone on the platform will benefit from the added freedom of wireless, consider the Mobility Test before providing each musician and singer with a system of his or her own.

Our recommendation is that anyone who is assigned a fixed position on the platform (such as drummers, keyboard players and choir members) be provided with wired microphones. While others that will want the ability to move around on stage or beyond.

Wireless for others

Consider the needs of each member - is mobility a must-have?

Microphone Recommendations for the Pastor

The pastor's audio quality is the most important concern for any house of worship audio system. Any of the following microphones will work for the pastor:

1. A headset microphone with a bodypack transmitter (top recommendation)

2. A lavalier microphone with a bodypack transmitter

3. A handheld microphone with a built-in transmitter

Why the Headset Is the Best Recommendation

The closer you can position the microphone to the sound source, in this case the pastor’s mouth, the better.

A lavaliere microphone is usually attached to the robe or lapel, which positions the microphone a few inches away from the sound source and not in the sound’s direct path. For this reason, the sound is not as clear and becomes softer and louder when the pastor looks from side to side or up and down.

A headset microphone allows you to position the microphone right at the pastor’s mouth or jaw line. When the pastor looks left or right – or even swivels to look behind – the microphone stays positioned at the mouth and the sound level remains the same.

It also enables higher gain-before-feedback. This lets you increase the pastor’s volume level – as needed of course – with less risk of feedback. Since placing microphones as close to the sound sources as possible is the best way to avoid feedback, a headset microphone is a better choice for this reason than a lavaliere.

Many pastors might object to the headset microphone for aesthetic reasons. Luckily headset microphones now come in a variety of colors and profiles. Try to match the microphone to the pastor's skin color to make them less visually distracting to the congregation.

If you need to convince the pastor or others involved, you might want to try a simple sound test: make recordings of two rehearsals, one using a lavaliere microphone and one using a headset. When you play the recordings back, the pastor should hear the dramatic difference in sound clarity and consistency and can then decide just how much sound quality is being traded for aesthetics.

Best Practices for Improving the Audio

The following are general guidelines and considerations for improving the audio quality in any live reinforcement application.

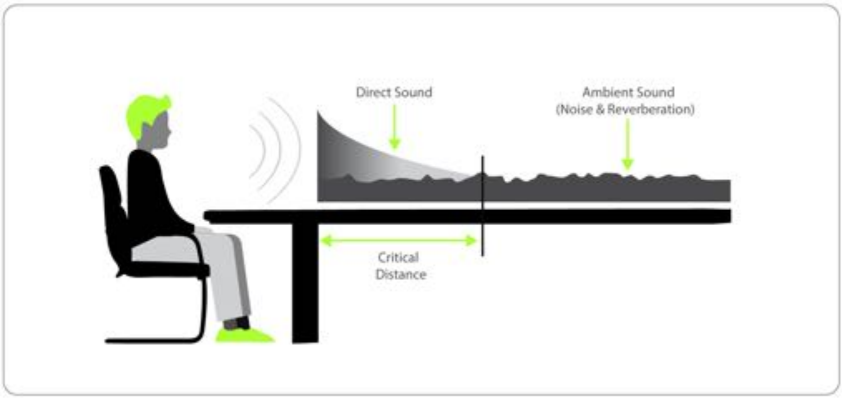
Limit Microphones in Use

A 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 only one microphone!

Microphone Placement

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

Recall that sounds can be categorized as desired or undesired and that the sound field, or 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. Critical distance is determined by the loudness of the direct sound relative to the loudness of the ambient sound. A quiet talker in a noisy room has a short critical distance while a loud talker in a quiet room has a longer critical distance. In practice, microphones must be placed much closer than the critical distance to get an acceptable ratio of direct-to-ambient sound.



Critical Distance

Keep microphones close to the sound source for more direct signal, reducing ambient and undesired sounds.

This brings up the misconception of “reach”, or distant pickup capability. There is no such thing as a microphone reach, it is simply the proportion of direct vs. ambient sound picked up by a microphone. This 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 for a hypercardioid (2 times greater).

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 to achieve the same output level. However, for a very weak source, or a very high ambient sound level, the acceptable omni location (again, less than the critical distance) could be as little as 3 inches away, for example. In this case, even a hypercardioid could only be used 6 inches away.

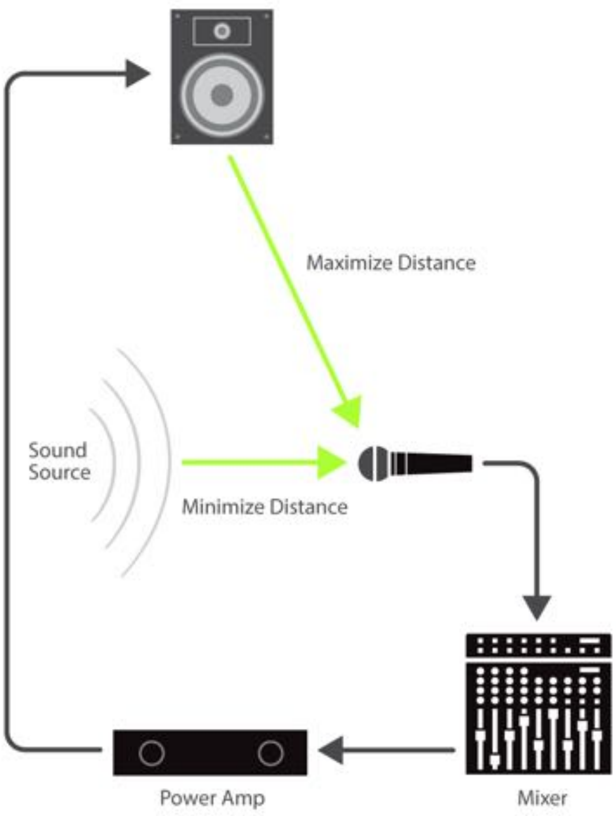
To put it another way: directional microphones are not more sensitive to on axis sound - they are just less sensitive to off-axis sound.

Feedback

In the normal operation of a sound system, some of the sound produced by the loudspeakers is picked up by the microphone and re-enters the system. As the gain of the system is increased, the level of the sound from the loudspeakers at the microphone also increases. Eventually, at some gain setting, this re-entrant sound will be amplified to the same level as the original sound picked up by the microphone. At this point the system will begin to “ring” or oscillate. Higher gain will result in the sustained “howl” or drone known as feedback.

There are many factors that affect the *potential acoustic gain* (maximum gain-before-feedback) of a sound system. By far, the most important ones are the relative distances between the sound source and the microphone, between the microphone and the loudspeaker, and between the loudspeaker and the listener. The number of “open” or active microphones also plays a strong role. These factors are discussed in Potential Acoustic Gain.

Lesser factors are the directional characteristics of the microphones and loudspeakers, local acoustical reflections, room reverberation, and the overall frequency response of the sound system. Use of directional microphones and directional loudspeakers can reduce the amount of direct sound picked up by the microphone from the loudspeaker by aiming them away from each other. Of course this is limited by the directional or “pattern” control of the devices.



Acoustical and Electrical Feedback Path

In practice, loudspeakers have very little directivity at low frequencies (where the wavelength is large compared to the speaker size).

Acoustic reflections from objects near the microphone can aggravate feedback problems. For example: sound from a monitor speaker placed behind the microphone can reflect off the performer's face into the front of the microphone, or a lectern surface can reflect the sound from an overhead cluster. Placing a hand on the front of or around the grille of a microphone can severely disrupt its polar pattern and frequency response.

Room reverberation increases the overall sound level throughout the room. Because it causes sound to persist even after the source stops, ringing and feedback tend to be more sustained. Since reverberation is not uniform with frequency it may also increase the likelihood of feedback at certain frequencies.

In fact, the overall frequency response of the sound system is affected by each component in the system as well as the room response. Feedback occurs first at the frequency that has the highest sensitivity in the system response curve. A peak in the response of a microphone or loudspeaker or an unusual boost in an equalizer can trigger feedback as the system gain is increased. Flat response systems can generally operate with more gain-before-feedback. Judicious use of equalizers can improve the stability of a sound system if feedback is occurring just at a few specific frequencies. However, equalizers will not allow the system to exceed the inherent limits of the PAG calculation.

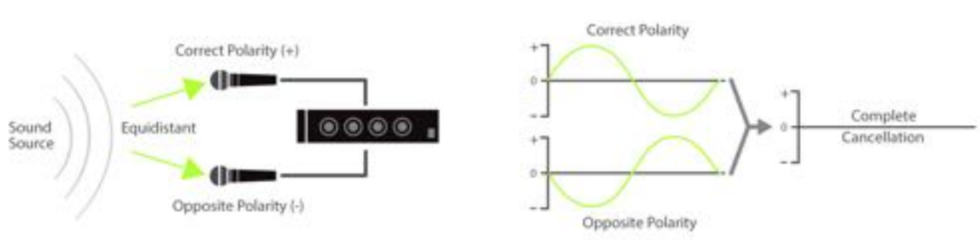
This leads to the first and most important rule of microphone placement: **Place the microphone as close as practical to the desired sound source**.

It has several corollaries: place the microphone as far as possible from loudspeakers and other undesired sources; use directional microphones to minimize ambient sound pickup; aim directional microphones toward the desired sound and/or away from undesired sound; and 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 a choir, may require greater distance, depending on the microphones' directionality; extremely loud sources may require greater distance to avoid overload of some sensitive condenser microphones; and close vocal 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 undesired sound, and the least likelihood of feedback.

Interference Effects

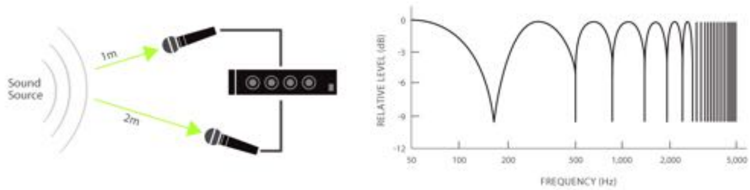
An important consideration in microphone use is acoustic interference. Interference effects may occur whenever delayed versions of the same sound are mixed together, acoustically or electrically. With microphones, this may happen in several ways: microphones of reverse polarity picking up the same sound,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

A severe loss of sound, especially low frequencies, occurs when a microphone with reverse polarity is placed next to another of correct polarity 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.

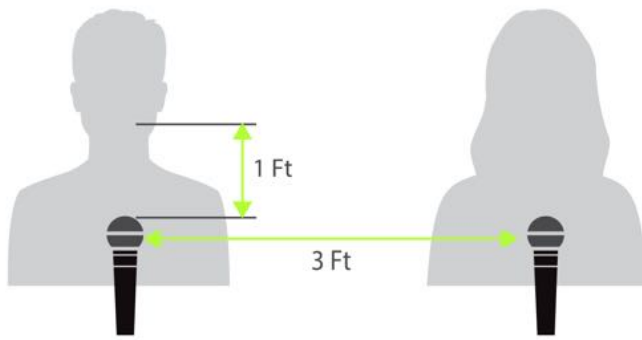


Although there is an international standard for microphone polarity (pin 2+, pin 3-), a reversal may be found in an incorrectly wired microphone cable. It can be identified by checking each microphone and cable against a microphone and cable that are known to be correct. **In any installation, all microphones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

The second form of interference is the result of multiple microphone pickup and 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 near 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 distances between the microphones. This effect is called “comb filtering” because the resulting frequency response curve resembles the teeth of a comb. As the delay time increases, comb filtering starts at lower frequencies. It is especially noticeable at middle and high frequencies, and creates a “hollow”, distant sound.



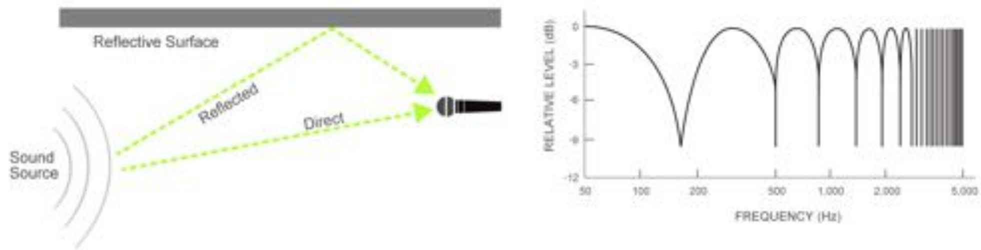
The solution to this problem is to use the three-to- one rule: **for multiple microphones, the microphone-to-microphone distance should be at least three times the source-to-microphone distance.**

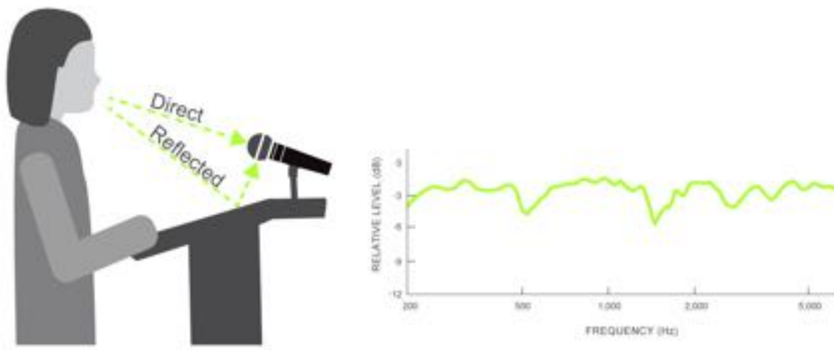


For example, when using individual microphones on a vocal group, if a singer’s microphone is one foot away, then the next nearest microphone should be at least three feet away from the first. This ensures that direct sound from the singer will not be strong enough to cause noticeable interference when picked up by the more distant microphones. As the source-to- microphone distance increases, the distance to adjacent microphones must also be increased.

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 sound applications: for area pickup applications, such as choir lofts and stages, each section or area should be covered by only one microphone; for lectern applications, only one microphone should be used; when a lavalier microphone wearer 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 worship facility settings: hardwood or stone floors, brick or glass walls, wood or plaster ceilings, and solid lecterns and altars. Recall that reflected sound is always delayed relative to direct sound. When the delayed, reflected sound arrives with the direct sound at the microphone, acoustic comb filtering is again the result.



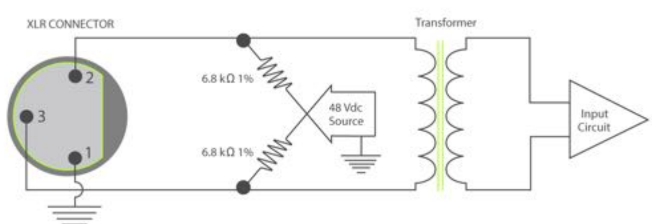


To minimize reflection pickup, **avoid using microphones near acoustically reflective surfaces**. If this is not possible, consider using a surface-mount microphone on the primary reflecting surface.

Microphone Connections

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 cables and connectors to achieve maximum 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 used for any permanent microphone installation**.



Phantom Power for Condenser Microphones

Phantom power, sometimes called “simplex”, is provided through the microphone cable itself. It is a DC (direct current) voltage that may range from 9 to 48 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 that 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 correctly hooked up to any standard phantom supply.

If a balanced microphone is incorrectly 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, an isolating transformer should be inserted. By the same token, it is also not possible to damage any standard phantom power source by improper microphone connection.

Good phantom power practices include:

- **Checking that phantom voltage is sufficient** for the selected condenser microphone(s)
- **Turning 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
- **Checking that microphones and cables are properly wired**

Following these practices will make condenser microphone use almost as simple as that of dynamics.

Phantom Power Versus Bias Voltage

In a condenser microphone, one function of the circuitry is to convert the very high impedance of the condenser element to a lower impedance. For an electret condenser (the most common type) this is done by a single transistor. Some condenser designs, such as lavalier types or miniature hanging types, have their electronics separate from the microphone element. In these models, the impedance converting transistor is built in to the microphone element itself. The main part of the circuitry is contained in a separate module or pack usually connected to the element by a thin shielded cable.

The main electronics of such designs operate on phantom power supplied through the microphone cable or by means of a battery in the pack itself. However, the impedance-converting transistor in the microphone element also requires power in a form known as “bias” voltage. This is a DC voltage, typically between 1.5 and 5 volts. It is carried on a single conductor in the miniature connecting cable, unlike phantom power which is carried on two conductors in the main microphone cable. In addition, the audio signal in the miniature cable is unbalanced while the signal in the main cable is balanced.

This distinction between phantom power and bias voltage is important for two reasons. The first concerns the use of wireless transmitters. Body-pack transmitters which operate on 9 volt (or smaller) batteries cannot provide phantom power (12-48 volts DC). This prevents their use with phantom-powered condenser microphones. However, the body-pack transmitter can provide bias voltage (1.5-5 volts DC). This allows a condenser microphone element with an integrated impedance-converting transistor to be used directly with a body-pack transmitter. Miniature condenser lavalier types as well as other designs which have separate electronics can be operated with wireless systems in this way.

The second reason concerns the wired installation of condenser microphones with separate electronic assemblies such as miniature hanging microphones for choir, congregation, or other “area” applications. Since the audio signal in the cable between the microphone element and the electronics is unbalanced, it is more susceptible to pickup of electronic noise. This is particularly true for radio frequency noise because the cable itself can act as an antenna, especially for a nearby AM radio station. For this reason it is strongly recommended to keep the length of this part of the cable as short as possible, preferably less than 35 feet. It is a much better practice to extend the length of the balanced cable between the electronics assembly and the mixer input.

Troubleshooting: Not Enough Gain Before Feedback?

Here's what you can do (in order of importance):

- Move microphones closer to sources
- Move loudspeakers farther from microphones
- Move loudspeakers closer to listeners
- Reduce the number of open microphones
- Use directional microphones and loudspeakers
- Eliminate acoustic reflections near microphones
- Reduce room reverberation by acoustic treatment
- Use equalizers to reduce system gain at feedback frequencies

Selecting the Right Wireless Systems

Start Small and Take the Mobility Test

Something you might realize fairly quickly is that this could get expensive.

Since you probably already have the audio systems you require for your day-to-day needs and daily services – no matter how good they might sound or look – adding new technologies will seem like a luxury, instead of a necessity.

If this is the case, you are not alone. Houses of worship have to make their limited budgets go farther than most other organizations. The good news is that the costs for these systems are coming down and the better news is that just a few systems can make a huge difference in the overall sound quality and aesthetics of your service.

Whether this is the praise leader, who can now wander freely around the platform, or the pastor, who can get closer to or into his congregation, you will have given them freedom and removed the cords that follow them around the platform – both of which will increase their connection to the congregation.

The Mobility Test

How do you decide which members of the praise team get both wireless microphone systems and wireless personal monitoring systems? Who gets only wired versions? And should any of them have a combination of the two?

A rule of thumb here is that it’s either all wired or all wireless. It is rarely, if ever, a combination of the two.

As you went through the members of the worship team determining who might need a wireless microphone system and then went through the same list deciding who might need a wired or wireless personal monitor, you should have seen that the same people who needed wireless microphones also needed wireless monitors.

The reason for this is ***mobility***.

Either they can benefit from mobility or they cannot. Either they will use their newfound freedom or they will stick to the same spot as before without much change.

If you have a young, rambunctious guitar player as part of your praise band, she will probably make very good use of her ability to move around without wires or losing her mix. If the bass player uses an upright bass or plays seated, perhaps a wired systems will do just fine.

A drummer, for example, will probably have no need for a wireless microphone system or wireless monitoring since he is in a fixed location and you can easily hide the cords that connect his systems to the mixer.

How to Select the Right Wireless Microphone System

There are many options for wireless microphones for church, from simple analog microphone systems to complex, multi-channel digital microphones systems. We understand that choosing the right wireless microphone system based on your unique needs is critical in amplifying the message.

Next, we will cover several critical factors to consider when selecting a wireless microphone system and provide wireless best practices for your House of Worship.

For further reading, the following link provides more information on selecting the right wireless system for Houses of Worship:

https://www.shure.com/en-US/performance-production/applications/houses-of-worship

For best results, discuss your requirements with a sound contractor or an applications specialist before making a final decision on your solution.

Audio Needs and Microphone Selection

Earlier we have shown the components of a wireless system and some of the set-ups that best fit the individuals who might be using them.

Now you need to identify the sound source you want to mic. Is it someone's voice? Is the person singing or speaking? Or is it an electric instrument like a guitar that plugs directly into the bodypack, or an acoustic instrument like a saxophone that needs an external microphone?

Count the number of users and/or rooms that might require any of the following configurations:

- Handheld microphone (with built-in transmitter)
- Headworn microphone with bodypack
- Lavaliere microphone with bodypack
- Clip-on microphone with bodypack
- Instrument cable with bodypack

This step is critical to determine the transmitter types and audio input requirements.

Location of Operation

One location – If you intend to use your wireless microphone system(s) in one location, you only need to make sure you select a system that operates on frequencies compatible with your location's TV channel frequencies.

Multiple locations – If you intend to use your wireless system(s) in different cities, you will likely encounter different active TV channels. Here, you should make sure your systems tune to frequencies compatible in the regions you are traveling to.

You should also want to consider mounting your equipment in a small rack case to make it easier to transport – especially if you are bringing more than one wireless system with you.

International – Very few wireless microphone systems work worldwide. If you are planning to use your wireless systems in foreign locations, you need to be even more careful about frequency selection and travel restrictions. It is best to rent or borrow systems in other countries.

visit www.shure.com/frequency to learn more about which frequency ranges are best for your requirements.

Do we need one system or many systems?

One system – if you are operating one system in a location where no other wireless systems are in use, then you will not have any multisystem needs to manage.

Multiple systems – If you plan to use more than one wireless system, you will need to carefully select frequencies to make sure that each system is compatible with the others. Also, there is a limit to the number of wireless systems that can be used in one location, which brings us to the final consideration:

Features and Budget

The adage that you ‘get what you pay for’ holds true with wireless systems. While the prices have come down and the features have improved, you still need to weigh your budget against your needs – especially when you are buying multiple systems for one location.

Better wireless systems allow you to operate more units at the same time without interference and are able to operate across larger bands of frequencies.

The key to any wireless system is the confidence you have in its ability to provide sound clarity that rivals its wired cousins. Your need for user-friendly features to locate open frequencies, avoid dropouts, and get clear consistent sound has not gone unnoticed by the manufacturers of these systems. More and more wireless systems are now including increasingly sophisticated technologies, such as ‘autoscan’ and ‘Audio Reference Companding,’ to help users get the sound and signal they want without having to worry about the technical issues. Before making any major system purchases, you might want to spend a little time researching the latest features and comparing their costs and benefits to your needs and budget.

Three Applications Where Wireless Is Key

Drama Productions

Drama productions, skits, and other presentations are becoming more and more part of the overall worship. In these cases, wireless microphone systems and wireless personal monitoring systems play a key role:

- Let people speak more naturally, allowing the audience to hear all the subtleties of the dialogue.
- Blend in with even the simplest costumes to provide clear audio with no distractions.
- Allow the actors to discreetly hear the director's feed or music cues – and even dialogue prompts during the performance – without the knowledge of the congregation.
- Clean up the overall sound in the house, since there is less need for monitor loudspeakers.

Place microphones and bodypack transmitters in wigs, costumes, props, or other items on stage to hide them from the audience.

Conceal the Bodypack in a Wig

The ADX1M transmitter is designed for wearing close to the body. It is perfect for theatrical or other performances when wearing the transmitter for extended period of time under clothing or a costume.

Portable Churches or Off-site Services

Your wireless microphone systems and personal monitoring systems can quickly turn even the sparsest room into a great worship space.

Since you have fewer cables to tape down, no bulky wedges to carry, and the mixes and levels can all be preset, you can be set up and start the service far faster than you might expect. You can also tear down quickly, allowing you more time on the service and less time packing up.

Portable Kit

A portable rig consisting of a wireless system (shown) and a few personal monitoring systems can work for axillary presentation areas.

One praise leader for a ‘portable church’ describes his solution as follows:

“I have a small rack with all the systems in it preset. I have a briefcase with all of our wireless mics and monitors in it. That’s all I need. And I know we’ll get a consistent mix right out of the briefcase.

All the time we used to spend setting up has been eliminated and there is almost nothing to unload from the van or load back up once we are at our time limit.

When you’re a portable church and you’re paying for the room by the hour, every minute you save is huge to us and our worshippers.”

Christian Touring Rock Bands and Traveling Praise Bands

For bands that travel, personal monitoring systems are becoming more and more common. Beyond the ability to clearly hear the desired mixes wherever you are on the stage or platform, your band will also realize the following advantages:

- far less equipment to transport
- faster set-up
- more consistent mixes
- certainty that you will always have enough monitor mixes (by use of an optional monitor mixer)
- less reliance on the skills or availability of the on-site monitor or sound engineer

Additionally, by using wireless systems throughout, you can work with nearly any space without fear of adding cables to an already crowded area or having to wait for the entire performance to end before collecting all your gear.

All in all, with the combination of wireless microphone systems and personal monitor systems, you put the control in your team’s hands and leave less to chance.

Touring Rig

A portable rig consisting of a wireless system (shown) and a few personal monitoring systems can work for axillary presentation areas.

Selecting the Right Personal Monitor System

When selecting personal monitoring systems for your house of worship, you need to answer all of the following questions:

1. **How many people will be using monitors?**
2. **Will the users be stationary or will they want to move freely around the platform?**
3. **Can they share monitor mixes or will they need to have their own?**
4. **Stereo or mono?**
5. **What is the best use of your budget?**

Answering these questions correctly, and fully, before purchasing personal monitor systems will help assure you have the flexibility to meet the widest variety of services and that you have used your budget most wisely. With that in mind, let's look at each of these questions individually:

How many people need monitors?

First, consider the role and needs of all of the people on the platform. Then decide if a personal monitor makes sense for each of these members of your worship ministry.

The pastor – can benefit from being certain his or her message is heard more clearly; can speak and hear at levels that are more comfortable and natural; has the ability to receive discreet cues and other information; can choose which mixes he or she wants to hear.

The praise leader – can hear her band and/or her instrument with no distractions; can sing at a level that is comfortable and more natural; can receive cues from the pastor or off-platform, as well as give her own cues back; can select what other mixes she wants to hear.

The members of the praise band, including guitarists, drummer, bass player, background vocalists, keyboardist, etc. – will be able to hear their own sounds without the 'volume war' associated with floor monitors; can hear the click track directly; can receive cues from the praise leader.

You probably already know which members of your praise band are the biggest 'volume war' offenders. This could be the best place to start when trying to get your team to adopt personal monitors.

The choir leader – all the same benefits as the praise leader. Choir leaders most commonly use personal monitors to hear the blend of the choir in the background.

The lead singers or the choir soloist – can sing at a level that is comfortable and more natural; can receive cues from the praise leader; can select what other mixes to hear; can stop worrying about echo or reverberation.

Since choir members do not commonly monitor their sound, there is rarely a need to consider giving anyone except the soloist a personal monitor system.

Audio/Tech engineer – will also find many uses for a personal monitor system. A great technique that is employed by many audio engineers is using in-ear monitors to select the right spot for microphone placement, especially for room miking. When listening to the microphone with in-ear monitors, the audio engineer will hear only what the microphone hears and none of the reflections from walls or other obstructions. This makes selection of the best locations for microphones an easier and more accurate process. This is also useful when placing microphones in front of loud instruments like guitar amps and kick drums. The engineer can walk right to the front of the amp cabinet with a microphone and position the microphone for the best sound – without being exposed to the louder than normal sound pressure levels.

Will the users be stationary or will they want to move freely around the platform?

Now that you have counted the number of people who might need personal monitors, determine whether or not they need to move freely around the platform. This will help you decide whether they can use wired personal monitors or if they might need the mobility of a wireless personal monitor.

A good rule of thumb is as follows:

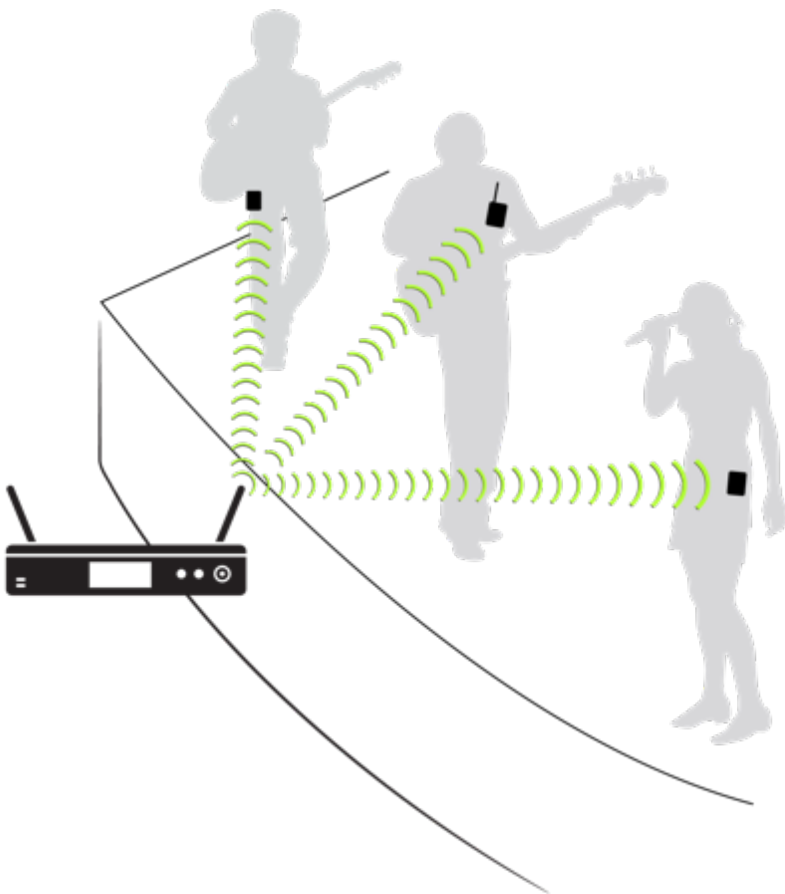
- Get wired versions for the drummer, keyboardist and back-up singers – all will likely remain in a fixed place during the service.
- Choose wireless versions for the praise leader, guitarists, the pastor and any soloists – as they will benefit more from the freedom to move about the platform. It is also a good idea to get the choir leader a wireless system, since he or she is standing in a place where cables might cause other ministry members to trip as they go by.

Can they share monitor mixes or will they need to have their own?

First determine how many mixes are presently used and if this number of mixes would suffice.

Then, alongside where you noted whether each user required a wired or wireless version, make an additional notation. This one is for whether they can share the overall mix or might need to have a personal mix.

Shared mix – Everyone sharing a monitor mix will be listening to the same exact mix. So long as they can all agree, sharing a mix is an easier and more cost-effective way of providing ‘in ear’ monitoring for a larger number of people.



One Transmitter for Multiple Users

A single personal monitor transmitter can send the same mix to multiple receivers.

When two or more users are able to share the same mix, you need only give them bodypack receivers and can use a single wireless transmitter to send this mix to them all. For example: If a few members of a praise band want to hear the same mix, they can utilize the same wireless transmitter. This will let you provide two or more users with wireless monitoring for a lower cost than if they all have their own transmitter.

Remember: Every mix requires its own transmitter, but each person who monitors that mix only requires a receiver.

Personal mix – The situation may occur, where each user has their own requirements for monitoring. Everyone wants to hear a mix that's a little different from the rest. Each unique mix would require a dedicated transmitter.

If your current mixing console does not distribute enough mixes ("auxes") to support all the mixes you need, you might reconsider sharing mixes. The other options would be investing in a dedicated monitor mixer, or upgrading your console.

Mono or stereo … or more?

In *mono*, both earphones reproduce the same audio. Not optimal, certainly, but often this is the lowest cost solution.

In *stereo*, the earphones produce the fullest, most accurate monitor sound available. These include both a Left and a Right signal – just like your CD player and stereo system – and also enables lower listening levels by separating sounds spatially instead of purely by volume. For example, this lets a praise band with two guitar players hear one guitar in the left ear and the other guitar in the right ear, creating a more realistic listening environment. Also: If you include the congregation as one of the mixes, a stereo system will allow you to hear them more naturally, with sounds from the left side of the house coming through to your left ear, for example. In short: *If you can afford stereo, which most systems now provide, it is well worth the additional cost.*

You need to make sure your present mixing console has the ability to transmit a stereo mix, or you might not be able to use this feature.

- dual-mono; in which one stereo transmitter sends out two monaural mixes. In this case the users decide which mix to hear by using the pan control on their receiver.
- MixMode®; which is a proprietary Shure solution that allows you to hear two separate signals (such as a vocal and a band mix; or the band and a discreet communications channel) in both ears. With MixMode, the user can control the 'blend' or the relative volume levels of these two mixes with the balance (pan) control.

Determine your budget

As with any purchasing decision, the amount you can afford to spend becomes a factor in what product features are 'need to haves' and which are 'nice to haves.'

Fortunately, the increased popularity of personal monitor systems has resulted in a wider variety of options to meet nearly any budget. Also, you can easily upgrade and add systems over time. (See Chapter V, "Start Small.")

Shure Product Selection Charts

Microphones

Application	Model	Operating Principle	Frequency Response	Directionality	Physical Design
Lectern	MX400 Series	Condenser	Shaped, vocal	Cardioid/Supercardioid	Miniature gooseneck
Altar	MX300 Series	Condenser	Flat	Cardioid/Supercardioid	Boundary
Vocal	KSM9	Condenser	Shaped, vocal	Supercardioid/Cardioid	Handheld
	KSM8	Dynamic	Flat, vocal	Cardioid	Handheld
	Beta 87A/C	Condenser	Shaped, vocal	Supercardioid/Cardioid	Handheld
	Beta 58A	Dynamic	Shaped, vocal	Supercardioid	Handheld
	SM87	Condenser	Shaped, vocal	Supercardioid	Handheld
	SM58	Dynamic	Shaped, vocal	Cardioid	Handheld
	SM86	Condenser	Shaped, vocal	Cardioid	Handheld
Headworn	DH5	Condenser	Flat and Shaped, vocal	Omni	Headworn
	PGA31	Condenser	Shaped, vocal	Cardoid	Headworn
	MX153	Condenser	Shaped, vocal	Omni	Headworn
Lavalier	MX183	Condenser	Shaped, vocal	Omni	Miniature Lavalier
	MX184	Condenser	Shaped, vocal	Supercardioid	Miniature Lavalier
	MX185	Condenser	Shaped, vocal	Cardioid	Miniature Lavalier
Choir	DL4	Condenser	Flat and shaped, vocal	Omni	Sub-miniature lavalier
	MX200 Series	Condenser	Shaped, vocal	Cardioid/Supercardioid	Miniature overhead
	SM81	Condenser	Flat, variable	Cardioid	Full size
	SM137	Condenser	Flat	Cardioid	Full size
	KSM137	Condenser	Flat, variable	Cardioid	Full size
	PGA81	Condenser	Flat	Cardioid	Full size

Application	Model	Operating Principle	Frequency Response	Directionality	Physical Design
Congregation	MX200 Series	Condenser	Shaped, vocal	Cardioid/Supercardioid	Miniature overhead
	PGA81	Condenser	Flat	Cardioid	Full size
Acoustic Guitar	Beta 57A	Dynamic	Shaped, inst.	Supercardioid	Full size, stand mount
	KSM137	Condenser	Flat, variable	Cardioid	Full size, stand mount
	PGA81	Condenser	Flat	Cardioid	Full size
Piano	SM81	Condenser	Flat, variable	Cardioid	Full size
	SM94	Condenser	Flat	Cardioid	Full size
	Beta181 Series	Condenser	Flat	Various	Small side-address
	KSM137	Condenser	Flat, variable	Cardioid	Full size
	PGA81	Condenser	Flat	Cardioid	Full size
Organ	SM81	Condenser	Flat, variable	Cardioid	Full size
	KSM32	Condenser	Flat, variable	Cardioid	Full size
	KSM137	Condenser	Flat, variable	Cardioid	Full size
	PGA81	Condenser	Flat	Cardioid	Full size
Stage	MX300 Series	Condenser	Shaped, vocal	Cardioid/Supercardioid	Boundary, floor mount
	MX200 Series	Condenser	Shaped, vocal	Cardioid/Supercardioid	Miniature overhead
Stereo	VP88	Condenser	Flat, variable	M-S Stereo	Full size, stand mount
	SM81 (pair)	Condenser	Flat, variable	Cardioid	Full size, stand mount
	Beta181 Series (pair)	Condenser	Flat	Various	Small side-address
	KSM32 (pair)	Condenser	Flat, variable	Cardioid	Full size
	KSM137 (pair)	Condenser	Flat, variable	Cardioid	Full size

Wireless Systems

Features	BLX® Wireless Systems ANALOG	GLX-D® Wireless Systems DIGITAL	SLX-D® Wireless Systems DIGITAL	QLX-DTM Digital Systems DIGITAL	ULX-D® Digital Systems DIGITAL
Transmitter Configurations	Handheld, Headworn, Lavalier, Guitar/Bass, Instrument	Handheld, Headworn, Lavalier, Guitar/Bass, Instrument	Handheld, Headworn, Lavalier, Guitar/Bass, Instrument	Handheld, Headworn, Lavalier, Guitar/Bass, Instrument	Handheld, Headworn, Lavalier, Guitar/Bass, Instrument
Receiver Options	Single, Half-Rack, Dual	Single, Guitar Pedal	Single, Dual	Single	Single, Dual, Quad
Compatible Systems Per Band*	Up to 12	Up to 8	Up to 32	Up to 60	Up to 60
Compatible Systems Using Multiple Bands	23	Up to 4 Typical; 8 Maximum	61	129	149
Selectable Frequencies	Group Channel Only	Group Channel Only	Tune in 25KHz	Tune in 25KHz	Tune in 25KHz
Auto Setup Features	Quickscan	LINKFREQ	Guided Freq Setup, Channel Scan, Group Scan, Sync	Scan / Group Scan / Sync	Scan / Group Scan / Sync
Audio Reference Companding	Yes	No	N/A	N/A	N/A
Dante	No	No	No	No	Yes
Audio Summing	No	No	No	No	Dual and Quad Only
Furnished Antennas	BLX4/BLX88 = internal BLX4R = removable 1/4 wave	GLXD4 =attached 1/4 wave GLXD6 = Internal Monopole	Detachable 1/4 wave	Detachable 1/2 wave	Detachable 1/2 wave
Receiver Networking	No	No	Yes	Yes	Yes (Single) Yes (Dual and Quad)

Features	BLX® Wireless Systems ANALOG	GLX-D® Wireless Systems DIGITAL	SLX-D® Wireless Systems DIGITAL	QLX-DTM Digital Systems DIGITAL	ULX-D® Digital Systems DIGITAL
Computer Monitoring & Control	No	No	**Yes**(WWB6 only)	**Yes**(WWB6 only)	**Yes**(WWB6 only)
Rack Hardware	Optional (URT2) Half Rack: Included	Optional (URT2)	Included	Included	Included
Transmitter Display	Battery: BiColor LED G/CH: 7 Segment LED	Tri-Color LED	Backlit LCD	Backlit Multi-function LCD	Backlit Multi-function LCD
Encryption	No	No	Yes; AES-256	Yes; AES-256	Yes; AES-256
Batteries, Battery Life	2 AA > 14 hrs	**Rechargeable**: Lithium Ion Rechargeable up to 16 hrs	2 AA > 8 hrs	2 AA > 9 hrs (1 mW / 10mW RF power) **Rechargeable**: SB900 > 10 hrs	2 AA > 11 hrs (1 mW / 10mW RF power) or > 5 hrs (20 mW RF power) **Rechargeable** : SB900 > 12 hrs (1 mW / 10mW RF power) or 8 hrs (20 mW RF power)

*Actual number of compatible systems will vary depending on setup and environment

Mixers and Amplifiers

Features	SCM262	SCM268	SCM410	SCM800	SCM810
Transformer-balanced input		●			
Active-balanced input	●		●	●	●
Transformer-balanced output		●			
Active-balanced output	●		●	●	●
Low-Z mic-level input	●	●	●	●	●
Line level input	●		2	●	●
Aux level input	●	●		●	●

Features	SCM262	SCM268	SCM410	SCM800	SCM810
Mic level output	●	●	●		
Line level output	●	●	●	●	●
Phono jack aux level output	●	●	●		
Headphone output				●	●
Phantom power	●	●	●	●	●
48V phantom power				●	●
Peak meter		●	●	●	●
EQ	●		●	●	●
Linkable			●	●	●
Limiter			●	●	●
Stereo operation	●				
AC operation	●	●	●	●	●

1. From optional external adapter.

2. Internal modification or optional accessory..

Personal Stereo Monitor Systems

Features	PSM ® 300 with P3R	PSM®300 with P3RA	PSM®900	PSM®1000
Listening mode	mono, stereo, mix mode	mono, stereo, mix mode	stereo, mix mode	stereo, mix mode
Inputs	1/4” (6.3mm)	1/4” (6.3mm)	2 XLR/TRS, line level	2 XLR/TRS, line level
Split outputs	1/4” (6.3mm)	1/4” (6.3mm)	2 TRS, duplicate input	2 TRS, duplicate input
Frequency agile	yes	yes	yes	yes
Maximum compatible systems	15per band	15per band	20per band	39per band
Remote antenna options	yes	yes	yes	yes
Networkable	no	no	no	yes
Switchable Transmitter power	no	no	yes	yes

Features	PSM ® 300 with P3R	PSM®300 with P3RA	PSM®900	PSM®1000
Batteries/hours	AA/6	AA/6	2xAA/6	2xAA/6
Rechargeable option	no	yes	yes	yes

Streaming: Live Broadcast the Service

Historically, house of worship activities have been intended for in-person participants. In the pre-internet era, many churches provided limited participation for non-attendees, typically by making audio recordings of services that have been distributed in various formats. At the same time, larger congregations have used traditional radio and television broadcast technologies to produce worship service programs that could be received either in real-time or recorded for later viewing or listening.

Now with widespread access to the internet, it has become possible for nearly any house of worship to stream services in real-time or to stream recorded services that can be viewed on-demand. Since a worship service has visual and audible elements, capturing or streaming a service involves both video and audio production techniques. While these two aspects are eventually handled together in the final program, our focus will be on the audio portion of the production with reference to video only as needed.

How Does Streaming Work?

There are many streaming platforms or services that may be used. Some are accessible via mobile devices as well as via computers, but ultimately must connect to the internet for distribution. Although we will not consider the details of these services here, we note that the quality, reliability, and cost of these services varies significantly. However, the end product is largely dependent on the source material and how it is input. We also note that the audio part of the material, in particular the spoken word, is generally the more important in house of worship applications.

Audio production for streaming involves microphones, mixing, and some means of delivering that mix to the streaming platform. Microphone choice and placement for various sound sources is discussed elsewhere in this booklet but mixing and connection for streaming will be covered in this chapter.

Note that the “capture” of video and the “capture” of audio involve very different technologies that are required by the different principles of behavior of light versus behavior of sound. An important result of these differences is that the optimum position for capturing sound (the microphone location) is rarely (if ever) the optimum position for capturing the visual image (the camera location). The best microphone location(s) can only be achieved independently of the best camera location(s).

We will consider two streaming scenarios: streaming using an existing sound system and streaming without an existing sound system. The first scenario may seem somewhat more complex, but it will yield the best audio result. The second scenario may seem somewhat simpler but will generally be limited in audio quality unless additional equipment is used.

Scenario 1: Streaming With Your Existing Sound System

The typical house of worship sound system should already be capable of picking up the sound sources that need to be amplified in the venue. These will generally include the principal presenter, secondary presenters, musical worship leaders, and musicians/singers. In some cases, a choir may also be picked up by microphones for sound reinforcement. In all cases, the mix of these sources is adjusted to provide the best sound quality for the in-person congregation.

The congregation will hear some of these sources primarily through the sound system as well as some sources directly, such as a large choir or a large pipe organ. The overall sound is strongly affected by the acoustics of the space as well as the quality of the sound system. If it is desired to stream the sound that is heard by the in-person listeners to remote listeners, it may be necessary use additional microphone(s) to capture those sources that are primarily heard directly. For example, if the choir is not normally reinforced through the sound system, one or more microphones may need to be positioned to pick up the choir for the streaming sound mix.

Another example is the pickup of the congregation itself. The congregation does not need to be amplified to hear itself, but if congregational singing is desired for the streaming mix for example, then appropriate microphones must be placed to pick up the congregation.

As noted, the in-person sound quality is affected by the room acoustics. The acoustic effects are primarily due to reflection of sound by various finishes and structures within the space. This produces a reverberant quality that is unique to that venue. The sound system mix consists primarily of close-up microphones on the various sources. If this mix is also used as the streaming mix, it will contain little of the acoustic characteristic of the space. To compensate, it is possible to use “ambient” microphones to pick up some of the overall room sound. Alternatively, artificial reverberation can be added to the streaming mix to achieve a more “live” characteristic.

It is very important that ambient microphone(s) and congregation microphones should only be used in the streaming mix, not in the sound reinforcement mix.

This scenario assumes that the sound system has the capability of creating at least two separate mixes: one for the in-person sound system and one for the streaming audience. Most modern audio mixers can create a “main” mix as well as one or more auxiliary mixes. These “aux” mixes are typically used to create mixes for monitor speakers, in-ear monitors, and “effects” (FX) such as reverb or delay. If the existing mixer has at least one available aux mix that can be dedicated for streaming, it is relatively straightforward to create a suitable streaming mix.

If the existing sound system mixer does not have an available aux mix for streaming or if it is preferred to generate the streaming mix from some other location, it may be possible to integrate a separate mixer to accomplish this. The additional mixer would need to have access to each of the necessary input channels through a microphone splitter or some other means of connecting those sources to multiple mixers. Digital mixers that have audio network capabilities can be easily connected together for this purpose.

In either case, the streaming mix should contain each of the sources that make up the main mix as well as those additional inputs such as the choir, the congregation, and ambient microphones if desired. It is usually sufficient for the streaming mix to be mono. However, a stereo streaming mix can be created from two mono aux mixes (one for Left and one for Right) or a single stereo aux mix if available.

If one person is responsible for both the in-person mix and for the streaming mix, it is strongly suggested to use sound-isolating headphones to listen to the streaming mix when needed, especially if this is being done in the same location as the in-person mix. If a separate mixer is used for the streaming mix, it may be located where it is acoustically isolated from the sound reinforcement area.

Again, the streaming mix may include additional input sources beyond what the “live” sound mix uses. Achieving proper balance of all sources for streaming may initially require more attention and will likely require some adjustment during the service.

Once the streaming mix is created, it is necessary to deliver that mix to the streaming platform of choice. Generally, this is accomplished by using an audio-to-USB interface. A typical model will have one (or two) audio inputs that accept a mono (or stereo) signal from a mixer. The input connectors may be XLR and/or ¼” phone jacks. Most of these devices can handle a microphone level signal or a line level signal. The choice of level is dictated by the output level of the streaming mixer: if the aux outputs of the streaming mixer are line level, then the interface must be configured for that signal level. If the interface is configured for mic level, a line level input will cause distortion. Conversely, if the mixer output is mic level and the interface is set for line level, the result will be low audio level and increased noise.

Many interfaces include a headphone output that allows listening to the mix directly. This is very useful if it is not possible to monitor the aux mix outputs on the source mixer. Again, sound-isolating headphones are recommended.

The output of the typical interface is a USB signal that may use a variety of connectors: micro-USB, USB-C, or USB-A. There are some legacy interfaces that use IEEE 1394 or “Firewire” connections. The USB output should be connected to a computer for best results. The computer will recognize the interface as an audio source and that source can be selected by most streaming platforms directly.

It is also possible to connect some interfaces to mobile devices such as iPhones, iPads, and Android phones. This may be attractive when the mobile device is also serving as the camera for the audio/video stream. However, due to the length limitation of USB cables (about 5m) the mobile device would have to be located within that distance to the audio-to-USB interface.

Of course, the same USB cable length limitation exists when the interface is connected to a computer. However, there is no practical limit on the length of audio cables between the streaming mixer and the audio-to-USB interface. If the location of the interface is dictated by the location of the computer or mobile device, it should be possible to run audio cables as needed from the streaming mixer to the interface.

Currently, there are mixers that have USB interfaces built in. In addition to the standard main and aux mix analog outputs, these devices also have a USB output that can connect directly to a computer or other streaming device that accepts a USB input. If the existing mixer is not sufficient for creating a suitable streaming mix, an additional mixer that has a USB capability might be considered. This would eliminate the need for a separate audio-to-USB interface. Of course, this USB mixer would have to allow for connection of all desired inputs for the streaming mix. As noted previously, this could be accomplished by a microphone splitter or possibly a digital audio network.

There are also interfaces that combine multiple audio and video inputs in a single device that can connect directly to a computer. This type of device may be useful when multiple video and audio sources are required with the convenience of a single interface.

Scenario 2: Streaming Without an Existing Sound System

If a sound system is not in place, or if it is desired to stream a service without using the existing sound system, it is possible to use a mobile device or a laptop computer running a streaming application of some sort. In this case, the mobile device (or computer) is typically functioning as the video capture device (built-in camera) as well as the audio capture device (built-in microphone). As noted earlier, the principal limitation of this approach is that the microphone location is now the same as the camera location.

If the main presenter is the only part of the service that is critical, it may be possible to position the device close enough to the subject to get a suitable video image and acceptable audio pickup. The result would likely be limited to a “talking head” shot with whatever sound quality can be achieved with the built-in microphone at a relatively close distance. There are accessory microphones that may improve the audio quality to some extent by offering a more directional pickup pattern, but the improvement is limited as long as they are attached to the device.

The main limitation is that audio pickup quality degrades quickly with distance from the microphone, especially in less-than-ideal acoustic spaces. If the space is quiet and non-reverberant, it may be possible to get acceptable audio up to about 1m from the mouth of the presenter. At any greater distance, or with poor room acoustics, the resulting audio will likely be noisy and have reduced intelligibility. The only recourse is to keep the microphone as close as practical to the presenter’s mouth.

In this presenter-only case, the simplest solution is to use a lapel microphone that can connect to the mobile device or to the laptop computer by an analog cable. Inexpensive lapel microphones that fit this description are readily available and most mobile devices can accept an external microphone directly or via a simple adapter. Such a setup allows the mobile device to be positioned as desired for the best video framing and still maintain close-mic audio quality. Again, the audio cable length of the microphone is not a limitation and can usually be extended as needed.

It may also be convenient to use a wireless microphone for this example. This allows the presenter to move about with no change in the audio quality. The output of most wireless lapel microphone systems can be easily connected to a mobile device (or computer) in a similar fashion.

As described previously, there are audio interfaces that can connect to mobile devices and to computers digitally. These interfaces can accept the signals from wired or wireless microphones and are often able to be powered by the device itself. The interface may also have a headphone output to monitor the signal.

If it is desired to accommodate other audio sources beyond a single presenter, it is likely that additional microphones will be required. This would involve a multichannel digital interface or a mixer. Many audio interfaces have two or more microphone inputs. While it is possible to connect the output of an analog mixer in the same way as a single analog microphone, it is more efficient to consider a mixer that has a built-in interface with an output that is suitable for the mobile device or computer.

If multiple audio sources are to be used, but there is no local sound reinforcement, then the same suggestions for the streaming mix of Scenario 1 become applicable. The only significant difference is the absence of the “in-person” mix consideration. While this simplifies the job of the sound person to some extent, it begins to resemble a “sound system without loudspeakers” in terms of the required gear. Ultimately, to achieve high audio quality streaming some sort of multiple microphone system is usually in order.

Considerations for Any Stream

Hardwired Internet Connection	There are some general considerations for successful streaming. One of the most important is to use a hard-wired internet connection if at all possible. While it is convenient to use Wi-Fi connected devices, no typical wireless connection will have the bandwidth or reliability of a wired connection. This may be particularly true in locations where the local Wi-Fi service is open to guests and not setup for high bandwidth operations. If your streaming program is expected to have HD video and high-quality audio in real-time, a gigabit speed network should be in place. If Wi-Fi connection is unavoidable, turn off guest access, use a password-protected link, and use a higher speed connection (usually 5GHz) if available. If the streaming program is sent from a computer (recommended), make sure the computer is not bogged down with other tasks during the streaming operation. The streaming computer should also have sufficient horsepower to handle HD video processing and other requirements of the streaming platform.
Latency	Conventional wireless microphone systems transmit an audio signal in “real time”. That is, there is negligible delay (“latency”) in the transmission. This includes all analog systems and most digital systems that operate in the VHF or UHF range. However, there are some digital wireless systems that use other radio technologies (such as Bluetooth) that may have significant latency. These are not suitable because they may create noticeable lip-sync issues in the resulting video.
Video connector	In the same manner that using a separate audio system allows the best audio for streaming, it is strongly recommended to use a DSLR (or video camera) that has a direct HDMI output to allow for the best video quality. Such a camera can be connected to the computer running the streaming platform using an HDMI-to-USB interface. If multiple cameras are desired, an appropriate video switcher can be used to choose between cameras as needed. If the church already has a video camera system for recording or broadcast, that video signal can be connected to the streaming computer using a suitable video interface.

Frequently Asked Questions

Whether you are new to audio or just need clarification on a covered topic, lets look at the frequently asked questions regarding audio systems.

Wireless Microphone Systems: FAQ

Is there anything specific to a House of Worship that might cause interference?

Maybe. The house of worship across the street or any other organization within 100 yards might also be using wireless microphone systems. These systems could be set on frequencies that interfere with yours. If you suspect they might be using wireless systems, you should ask them which frequencies they are using and avoid these when selecting your systems.

Additional sources of interference include:

- robes with a significant amount of metal threading
- digital devices or digital processors (such as CD or DAT players/recorders, DVD players, computers, Digital Signal Processors) located too near the wireless receivers

How many wireless systems can I use at one time?

This varies by frequency, model and manufacturer. While you can use a significant number of total wireless systems at the same time you need to be careful to coordinate the frequencies correctly. This is covered somewhat within this booklet, but it's best to contact a sound contractor, your audio representative, or the manufacturer if you want to use more than the number of microphones indicated by the model you choose.

How can I make sure that the multiple systems are not interfering with each other?

First, it is always a good idea to consult the manufacturer's guidelines for frequency selection. Second, you might want to perform a listening test. Turn all of the systems on at once.

Put the transmitters where they will be during the service.

Then take each transmitter and, while talking or singing, walk around the entire worship platform and even up into the back rows. You will then determine if there is any interference and check for dropouts at the same time.

Note that this will only help you determine if your own systems are compatible. Systems being used by nearby organizations might still cause interference. If you are unsure where the interference may be coming from, contact the manufacturer of your wireless system.

Can I mix and match wireless systems from multiple manufacturers?

Yes, but here again frequency coordination could be an issue. It's best to contact a sound contractor, your audio representative, or one of the two manufacturers directly before doing so.

Do wireless microphones increase the likelihood of feedback?

Not because they are wireless, but because of the ability to take the microphones places where the feedback might occur. (I.e. the pastor walking in front of a loudspeaker)

Can I bring my wireless microphone on the road with me? …to other houses of worship?

This is covered within this booklet, but the short answer is: maybe. It depends on the frequency at which your system is set. If it is on the same frequency as a local TV channel or another wireless system, you will have trouble. If you know you will need to travel before you purchase the system, you should consider one that is frequency agile or can automatically search for open frequencies.

Tell me straight. Don't wired microphones sound better and aren't they easier to use?

Many people believe wired microphones sound better than their wireless counterparts, but this gap has closed dramatically in recent years, especially with digital wireless systems. Additionally, most people now believe that the added mobility of wireless microphones more than offsets any perceived difference in sound.

Personal Monitor Systems FAQ

Can we try a Personal Monitor system before we buy one?

This is possibly the most common question we receive.

The answer is "yes," but it is unlikely you will get this answer at your local audio retailer, since they are probably not set up to do so. There is a good chance your sound contractor or audio representative can help arrange a trial. If not, contact the manufacturer directly. Often, they can help arrange a demonstration unit.

I have not used in-ear monitors before. Are they comfortable and easy to use?

You should get the hang of inserting and wearing the earphones after just a few rehearsals or services. Adding the congregation's participation to the mix, by use of ambient microphones, can help remove some of the isolation. Either way, as with any new technology, you will soon get used to them and will just as quickly wonder how you expressed your worship without them.

Do I need a transmitter for everyone using a personal monitor system?

No. In many cases, more than one user can share the same transmitter, so long as they listen to the same mix and each have their own bodypack receiver, and earphones.

What if some band members want a personal monitor and some don't?

For maximum benefit, it is recommended that all band members be 'in ear.' In situations where some band members resist or budget does not allow you to provide personal monitors to every one in the band, it's acceptable to dip your toe in the water when bringing in these new technologies.

Some people take more time adopting earphones and forcing them to use something that makes them uncomfortable might affect their worship… which is not a good result.

While giving personal monitor systems to only one or two members of your praise band will not remove all the cables and wedges from the platform (which is the ultimate goal) it *will* decrease the number of wedges you require and will certainly help lessen the 'volume wars.'

Eventually your more reluctant team members will begin to see the advantages of personal monitor systems and might even dive in themselves.

I only have one output available on the mixer and I use that for the wedges. How can I add personal monitors?

Most personal monitor systems can also be used as pass-though devices for other personal monitors or floor monitors.

Here's how to do it:

1. Connect the input of the personal monitor transmitter to the monitor or aux output of the mixer.
2. You can now connect the floor monitor amplifier to the loop outputs on the personal monitor transmitter. Or you could connect another personal monitor system. In fact, you can daisy chain as many of these together as you want.

This lets you maximize the soundboard's one output. Also, users can change the volume they hear in their ears without affecting the level of the sound going to the other monitors.

Can I use a reverb unit or some sort of digital processor on my in-ear mix?

Yes, of course. But note that you are adding a little more delay in the signal that could be an issue with timing. Try any set up before you use it live to make sure there are no issues.

We have a lot of stained glass windows in our church. Does this affect my choice of monitoring systems?

Yes, if you mean whether you select floor wedges or personal monitoring systems. Stained glass windows (or *any* glass windows, in fact) are some of many architectural details specific to houses of worship that can cause reverberation and acoustic issues. Personal monitoring systems can help decrease the overall volume, which helps clean up the sound in reverberant houses of worship.

Can personal monitoring systems help with recording and broadcast needs?

Yes. Directors and producers looking to capture the service on CD/DVD can benefit from personal monitors. You can also use the monitor mix to record your music.

Contact your sound contractor or the manufacturer for more details on the added value of personal monitors or personal monitoring in these situations.

Learn More

What is Sound?

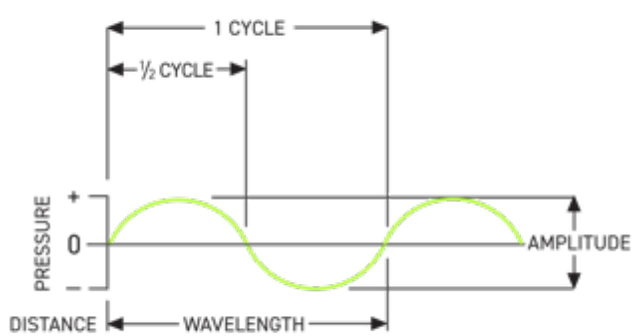
Introduction

Sound is produced by vibrating objects, such as musical instruments, loudspeakers, and, of course, human vocal cords. The mechanical vibrations of these objects push and pull the air which is immediately adjacent to them, causing a series of cyclic pressure changes that travel outward from the object, forming a pattern called a sound wave. *A sound wave is a series of pressure changes (cycles) moving through the air.*

Frequency and Amplitude

A sound wave can be described by its frequency, amplitude, and wavelength. The *frequency* of a sound wave is the rate at which the pressure changes occur, and determines the "pitch" of the sound. It is measured in Hertz (Hz), where 1 Hz is equal to 1 cycle-per second. The range of frequencies audible to the human ear extends from a low of about 20 Hz to a high of about 20,000 Hz.

The *amplitude* of a sound wave refers to the magnitude of the pressure changes and determines the “loudness” of the sound. Amplitude is measured in decibels (dB) of sound pressure level (SPL) 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 70 dB SPL. A change of 1 dB is about the smallest SPL difference that the human ear can detect, while 3 dB is a generally noticeable step, and an increase of 10 dB is perceived as a “doubling” of loudness. (See The Decibel)

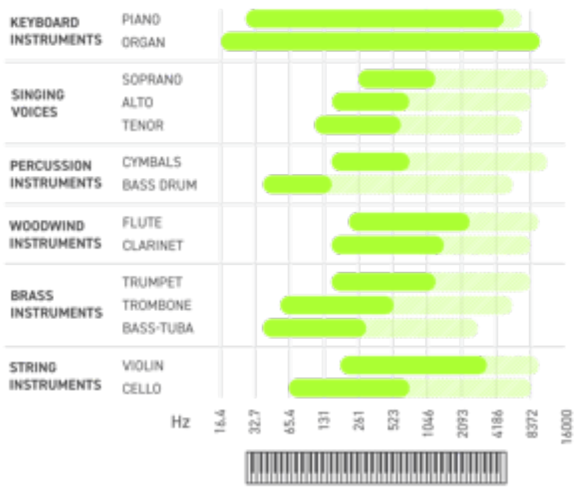


Schematic of a Sound Wave

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. The higher the frequency of sound, the shorter the wavelength, and the lower the frequency, the longer the wavelength.

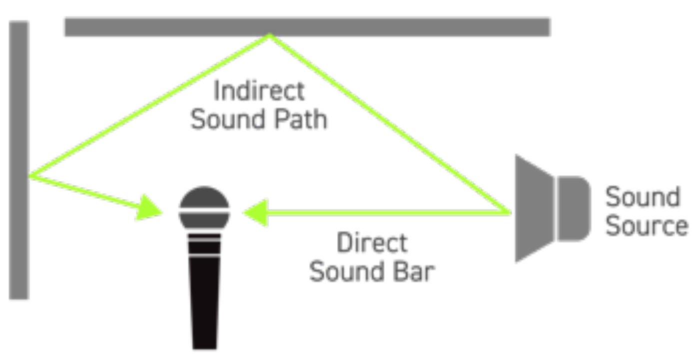
This corresponds to the sounds we hear in music. As a reference, the following chart shows which frequencies are produced by each instrument.



Instrument Frequency Ranges

Direct Sound Versus Indirect Sound

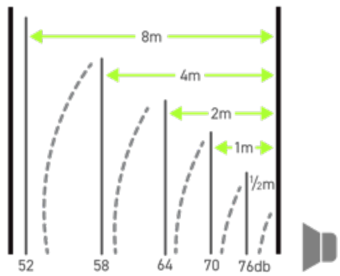
Direct sound travels from the sound source to the microphone (or the listener) in a straight line (the shortest path). *Indirect sound* is reflected by one or more surfaces before reaching the microphone (a longer path). Since sound travels through air at a constant speed (1130 feet per second or 343 meters per second), it takes a longer time for the indirect sound to arrive, and it is said to be "delayed" relative to the direct sound. Indoors, sound waves are reflected by the walls, ceiling, floor, and objects, so the microphone receives a mix of both direct and indirect sound.



Direct Versus Indirect Sound

A very important 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*.

Inverse Square Law



As the microphone moves way from the sound source, the signal spreads out over space and looses energy in any given direction, resulting in the predicable rate.

For example, when the distance from the source to the **microphone increases by a factor of two** (doubles), the sound level at the microphone **decreases by a factor of four** (the square of two). This results in a drop of 6 dB in sound pressure level (SPL), a substantial decrease. Likewise, when the distance from the mic to the source **is divided by two (cut in half), the sound level increases by 6 dB.**

In contrast, ambient sound, such as reverberation or room noise, has a relatively constant level everywhere in the room. Therefore, at a given distance from a sound source, the microphone will pick up a certain proportion of direct sound vs. ambient sound. **As the distance increases, the direct sound level decreases while the ambient sound level stays the same, so the ratio of direct to ambient sound decreases.** For speech intelligibility, the direct sound (speech in this case) should be at least 20 dB louder than the ambient sound at the listener's ear. This is called the *signal-to-noise (S/N) ratio*.

The Decibel

The decibel (dB) is an expression often used in electrical and acoustic measurements. The decibel is a number that represents a ratio of two values of a quantity such as voltage. It is actually a logarithmic ratio whose main purpose is to scale a large measurement range down to a much smaller and more useable range. The form of the decibel relationship for voltage is:

dB = 20 x log(V1/V2)

where 20 is a constant, V1 is one voltage, V2 is a reference voltage, and log is logarithm base 10.

Examples:

What is the relationship in decibels between 100 volts and 1 volt?

(dbV) dB = 20 x log(100/1)dB = 20 x log(100)dB = 20 x 2 (the log of 100 is 2)dB = 40

That is, 100 volts is 40dB greater than 1 volt.

What is the relationship in decibels between .0001 volt and 1 volt? (dbV)

dB = 20 x log(.001/1)dB = 20 x log(.001)dB = 20 x (-3) (the log of .001 is -3)dB = -60

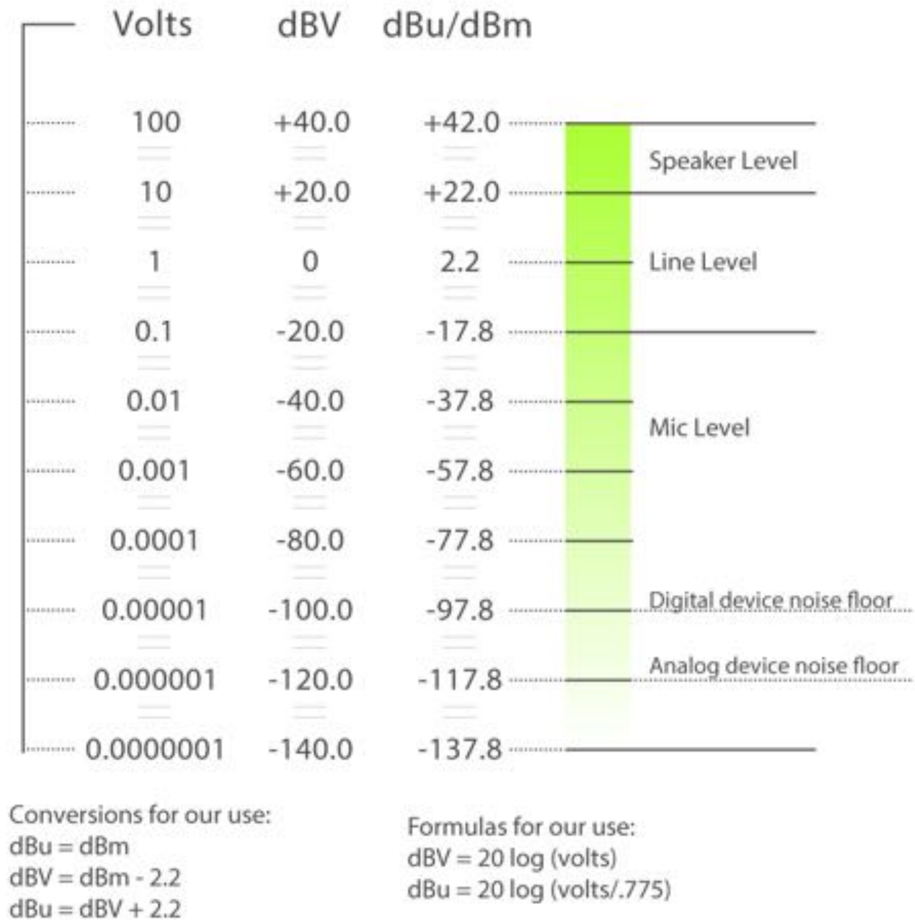
That is, .001 volt is 60 dB less than 1 volt.

Similarly: If one voltage is equal to the other, they are 0 dB different.

If one voltage is twice the other, they are 6 dB different.

If one voltage is ten times the other, they are 20 dB different.

Since the decibel is a ratio of two values, there must be an explicit or implicit reference value for any measurement given in dB. This is usually indicated by a suffix on the dB. Some devices are measured in dBV (reference to 1 Volt = 0 dBV), while others may be specified in dBu or dBm (reference to .775 V = 0dBu/dBm). Here is a chart that makes conversion for comparison easy:



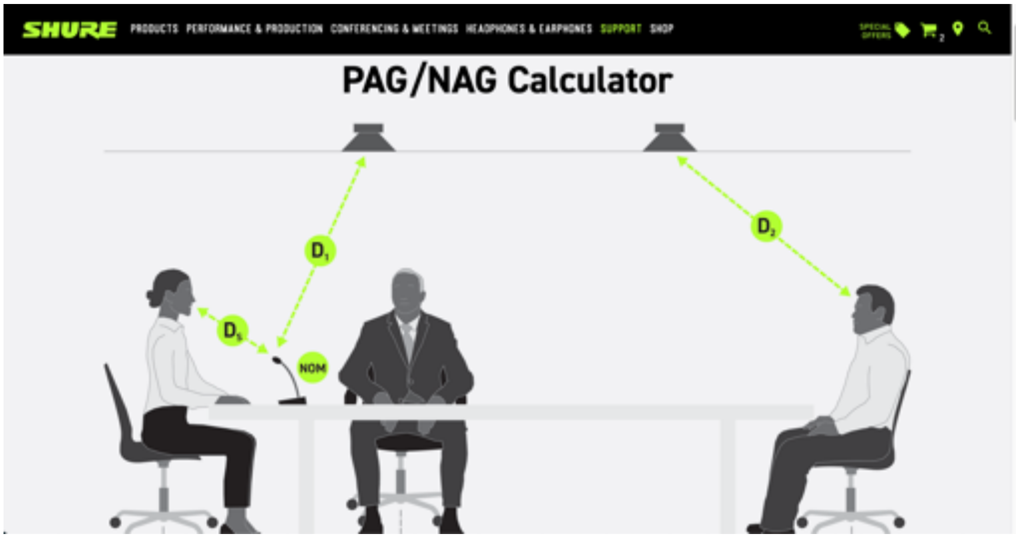
Conversion Chart

Audio equipment signal levels are generally broken into 3 main categories: Mic, Line, or Speaker Level. Aux level resides within the lower half of line level. The chart also shows at what voltages these categories exist.

One reason that the decibel is so useful in certain audio measurements is that this scaling function closely approximates the behavior of human hearing sensitivity. For example, a change of 1 dB SPL is about the smallest difference in loudness that can be perceived while a 3dB SPL change is generally noticeable. A 6 dB SPL change is quite noticeable and finally, a 10 dB SPL change is perceived as twice as loud.

Potential Acoustic Gain

Potential Acoustic Gain (PAG) vs. Needed Acoustic Gain (NAG)



The basic purpose of a sound reinforcement system is to deliver sufficient sound level to the audience so that they can hear and enjoy the performance throughout the listening area. As mentioned earlier, the amount of reinforcement needed depends on the loudness of the instruments or performers themselves and the size and acoustic nature of the venue. This Needed Acoustic Gain (NAG) is the amplification factor necessary so that the furthest listeners can hear as if they were close enough to hear the performers directly.

Visit the

PAG/NAG calculator on the Shure website, a free tool to easily make the calculations based on the specifics of your room and layout.

The solution can also be found by doing the math yourself:

To calculate NAG:

NAG = 20 x log (Df/Dn)

Where: Df = distance from sound source to furthest listener

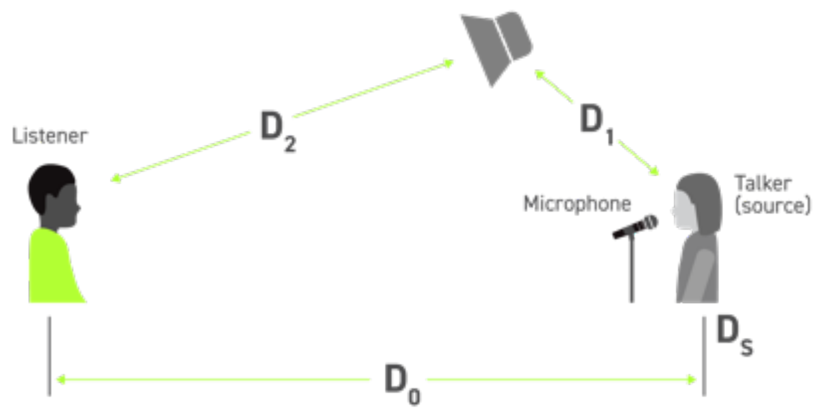
Dn = distance from sound source to nearest listener

log = logarithm to base 10

the sound source may be a musical instrument, a vocalist or perhaps a loudspeaker.

The equation for NAG is based on the inverse-square law, which says that the sound level decreases by 6 dB each time the distance to the source doubles. For example, the sound level (without a sound system) at the first row of the audience (10 feet from the stage) might be a comfortable 85 dB. At the last row of the audience (80 feet from the stage) the level will be 18 dB less or 67 dB. In this case the sound system needs to provide 18 dB of gain so that the last row can hear at the same level as the first row. The limitation in real-world sound systems is not how loud the system can get with a recorded sound source but rather how loud it can get with a microphone as its input. The maximum loudness is ultimately limited by acoustic feedback.

The amount of gain-before-feedback that a sound reinforcement system can provide may be estimated mathematically. This Potential Acoustic Gain (PAG) involves the distances between sound system components, the number of open mics, and other variables. The system will be sufficient if the calculated PAG is equal to or greater than the NAG. Following is an illustration showing the key distances.



Potential Acoustic Gain

The simplified PAG equation is:

PAG = 20 (log D1- log D2+log D0- log Ds)- 10 log NOM-6

Where: PAG = Potential Acoustic Gain (in dB)

Ds = distance from sound source to microphone

D0 = distance from sound source to furthest listener

D1 = distance from microphone to nearest loudspeaker

D2 = distance from loudspeaker to furthest listener

NOM = the number of open microphones

-6 = a 6 dB feedback stability margin

log = logarithm to base 10

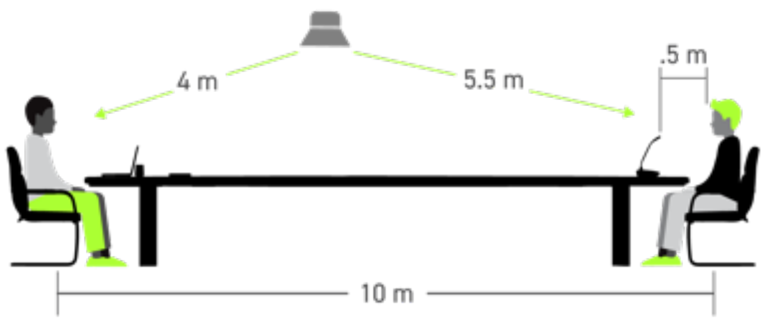
In order to make PAG as large as possible, that is, to provide the maximum gain-before-feedback, the following rules should be observed:

1) Place the microphone as close to the sound source as practical.

2) Keep the microphone as far away from the loudspeaker as practical.

3) Place the loudspeaker as close to the audience as practical.

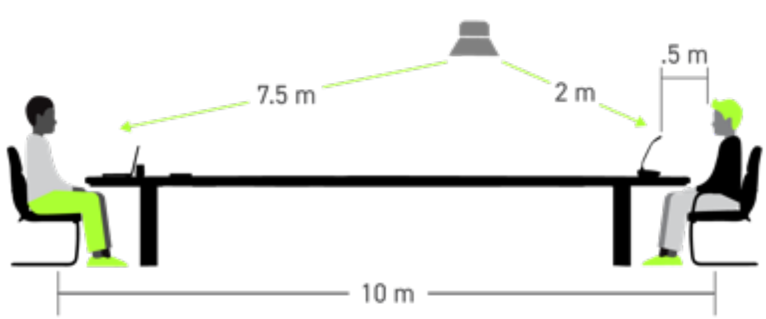
4) Keep the number of microphones to a minimum.



System will work: PAG>NAG

In particular, the logarithmic relationship means that to make a 6 dB change in the value of PAG the corresponding distance must be doubled or halved. For example, if a microphone is 1 ft. from an instrument, moving it to 2 ft. away will decrease the gain-before feedback by 6dB while moving it to 4 ft. away will decrease it by 12 dB. On the other hand, moving it to 6 in. away increases gain-before-feedback by 6dB while moving it to only 3 in. away will increase it by 12 dB. This is why the single most significant factor in maximizing gain-before-feedback is to place the microphone as close as practical to the sound source.

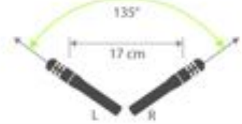
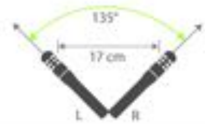
The NOM term in the PAG equation reflects the fact that gain-before-feedback decreases by 3 dB every time the number of open (active) microphones doubles. For example, if a system has a PAG of 20 dB with a single microphone, adding a second microphone will decrease PAG to 17dB and adding a third and fourth mic will decrease PAG to 14 dB. This is why the number of microphones should be kept to a minimum and why unused microphones should be turned off or attenuated. Essentially, the gain-before-feedback of a sound system can be evaluated strictly on the relative location of sources, microphones, loudspeakers, and audience, as well as the number of microphones, but without regard to the actual type of component. Though quite simple, the results are very useful as a best case estimate.



System will not work: PAG

Stereo Microphone Techniques

One exception to the rule about always erring on the minimum number of microphones, is stereo microphone technique, where two microphone elements are used to pick up the same sound source. These may be in the form of either separate standard microphones, or a single stereo microphone, combining the elements in one housing. In either case, the goal of stereo microphone application is to add the aspects of width and depth to the reproduced sound. This results in a more realistic image when heard through a stereo sound system. There are many techniques used to accomplish this goal, but they may all be categorized as follows: coincident, near-coincident, or spaced.

STEREO PICKUP SYSTEMS	MICROPHONE TYPES	MICROPHONE POSITIONS	
X-Y	2 - CARDIOID	AXES OF MAXIMUM RESPONSE AT 135° SPACING: COINCIDENT	
ORTF (FRENCH BROADCASTING ORGANIZATION)	2 - CARDIOID	AXES OF MAXIMUM RESPONSE AT 110° SPACING: NEAR-COINCIDENT (7 IN.)	
NOS (DUTCH BROADCASTING FOUNDATION)	2 - CARDIOID	AXES OF MAXIMUM RESPONSE AT 90° SPACING: NEAR-COINCIDENT (12 IN.)	
STEREOSONIC	2 - BIDIRECTIONAL	AXES OF MAXIMUM RESPONSE AT 90° SPACING: COINCIDENT	
MS (MID-SIDE)	1 - CARDIOID 1 - BIDIRECTIONAL	CARDIOID FORWARD-POINTED; BIDIRECTIONAL SIDE-POINTED; SPACING: COINCIDENT	
SPACED	2 - CARDIOID OR 2 - ONMIDIRECTIONAL	ANGLE AS DESIRED SPACING: 3-10 FT.	

Stereo Microphone Techniques

Coincident techniques use directional microphones, with the elements placed as close together as possible, but angled apart. The stereo image is a function only of the directional patterns of the microphones and the relative angle between them. This generally yields a stereo effect with modest “width” but good “localization” of sound sources. Single-housing/multi-element stereo microphones are also coincident types. They may contain unidirectional elements, bidirectional elements, or some combination of the two. Some of these, such as the M-S (Mid-Side) design, are capable of excellent (and sometimes variable) stereo width.

Since there is very little distance between coincident microphones, there is essentially no delay between the sounds picked up by them. This eliminates any potential interference (comb filtering) if the signals are combined for a monophonic sound system. Coincident techniques are thus mono-compatible

Near-coincident techniques also use unidirectional microphones, but they are placed with their elements 6 to 12 inches apart, and at some angle relative to each other. In this method, the stereo image is a function not only of directionality but also of distance. The result is good image width and accurate image localization. Since there is a finite distance between the microphones, and hence, some delay between the sounds picked up, there may be some noticeable interference effects if the signals are combined monophonically.

Spaced techniques may use unidirectional or omnidirectional microphones. They are placed 3 to 10 feet apart and may or may not be angled relative to each other. Here, the stereo image is primarily a function of the distance between the microphones, and not their directionality. This technique results in exaggerated stereo separation and somewhat indistinct imaging, and is primarily used to pick up the ambient sound of a space. Due to the large distance between microphones, severe interference effects may result when combining direct sounds in mono.

Glossary

3-to-1 Rule: When using multiple microphones, the distance between microphones should be at least 3 times the distance from each microphone to its intended sound source.

Absorption: The dissipation of sound energy by losses due to sound absorbent materials.

Active Circuitry: Electrical circuitry which requires power to operate, such as transistors and vacuum tubes.

Ambience : Room Acoustics or natural reverberation

.**Amplitude:** The strength or level of sound pressure or voltage.

Audio Chain: The series of interconnected audio equipment used for recording or sound reinforcement.

Backplate: The solid conductive disk that forms the fixed half of a condenser element.

Balanced: A circuit that carries information by means of two equal but opposite polarity signals, on two conductors.

Bidirectional Microphone: A microphone that picks up equally from two opposite directions. The angle of best rejection is 90 degrees from the front (or rear) of the microphone, that is, directly at the sides.

Boundary/Surface Microphone: A microphone designed to be mounted on an acoustically reflective surface.

Cardioid Microphone: A unidirectional micro phone with moderately wide front pickup (131 degrees). Angle of best rejection is 180 degrees from the front of the microphone, that is, directly at the rear.

Cartridge (Transducer): The element in a microphone that converts acoustical energy (sound) into electrical energy (the signal).

Clipping Level: The maximum electrical output signal level (dBV or dBu) that the microphone can produce before the output becomes distorted.

Close Pickup: Microphone placement within 2 feet of a sound source.

Comb Filtering: An interference effect in which the frequency response exhibits regular deep notches.

Condenser Microphone: A microphone that generates an electrical signal when sound waves vary the spacing between two charged surfaces: the diaphragm and the backplate.

Critical Distance: In acoustics, the distance from a sound source in a room at which the direct sound level is equal to the reverberant sound level.

Current: Charge flowing in an electrical circuit. Analogous to the amount of a fluid flowing in a pipe.

Decibel (dB): A number used to express relative output sensitivity. It is a logarithmic ratio.

Diaphragm: The thin membrane in a microphone which moves in response to sound waves.

Diffraction: The bending of sound waves around an object which is physically smaller than the wavelength of the sound.

Direct Sound: Sound which travels by a straight path from a sound source to a microphone or listener.

Distance Factor: The equivalent operating distance of a directional microphone compared to an omnidirectional microphone to achieve the same ratio of direct to reverberant sound.

Distant Pickup: Microphone placement farther than 2 feet from the sound source.

Dynamic Microphone: A microphone that generates an electrical signal when sound waves cause a conductor to vibrate in a magnetic field. In a moving-coil microphone, the conductor is a coil of wire attached to the diaphragm.

Dynamic Range: The range of amplitude of a sound source. Also, the range of sound level that a microphone can successfully pick up.

Echo: Reflection of sound that is delayed long enough (more than about 50 msec.) to be heard as a distinct repetition of the original sound.

Electret: A material (such as Teflon) that can retain a permanent electric charge.

EQ: Equalization or tone control to shape frequency response in some desired way.

Feedback: In a PA system consisting of a micro phone, amplifier, and loudspeaker, feedback is the ringing or howling sound caused by amplified sound from the loudspeaker entering the microphone and being re-amplified.

Flat Response: A frequency response that is uniform and equal at all frequencies.

Frequency: The rate of repetition of a cyclic phenomenon such as a sound wave.

Frequency Response Tailoring Switch: A switch on a microphone that affects the tone quality reproduced by the microphone by means of an equalization circuit. (Similar to a bass or treble control on a hi-fi receiver.)

Frequency Response: A graph showing how a microphone responds to various sound frequencies. It is a plot of electrical output (in decibels) vs. frequency (in Hertz).

Fundamental: The lowest frequency component of a complex waveform such as musical note. It establishes the basic pitch of the note.

Gain: Amplification of sound level or voltage.

Gain-Before-Feedback: The amount of gain that can be achieved in a sound system before feedback or ringing occurs.

Gobos: Movable panels used to reduce reflected sound in the recording environment.

Harmonic: Frequency components above the fundamental of a complex waveform. They are generally multiples of the fundamental which establish the timbre or tone of the note.

Hypercardioid: A unidirectional microphone with tighter front pickup (105 degrees) than a supercardioid, but with more rear pickup. Angle of best rejection is about 110 degrees from the front of the microphone.

Impedance: In an electrical circuit, opposition to the flow of alternating current, measured in ohms. A high impedance microphone has an impedance of 10,000 ohms or more. A low-impedance microphone has an impedance of 50 to 600 ohms. Interference Destructive combining of sound waves or electrical signals due to phase differences.

Inverse Square Law: States that direct sound levels increase (or decrease) by an amount proportional to the square of the change in distance.

Isolation: Freedom from leakage; the ability to reject unwanted sounds.

Leakage: Pickup of an instrument by a microphone intended to pick up another instrument. Creative leakage is artistically favorable leakage that adds a “loose” or “live” feel to a recording.

Maximum Sound Pressure Level: The maximum acoustic input signal level (dB SPL) that the microphone can accept before clipping occurs.

NAG: Needed Acoustic Gain is the amount of gain that a sound system must provide for a distant listener to hear as if he or she was close to the unamplified sound source.

Noise: Unwanted electrical or acoustic energy.

Noise Canceling: A microphone that rejects ambient or distant sound.

NOM: Number of open microphones in a sound system. Decreases gain-before-feedback by 3dB every time NOM doubles.

Omnidirectional Microphone: A microphone that picks up sound equally well from all directions.

Output Noise (Self-Noise): The amount of residual noise (dB SPL) generated by the electronics of a condenser microphone.

Overload: Exceeding the signal level capability of a microphone or electrical circuit.

PAG: Potential Acoustic Gain is the calculated gain that a sound system can achieve at or just below the point of feedback.

Phantom Power: A method of providing power to the electronics of a condenser microphone through the microphone cable.

Phase: The “time” relationship between cycles of different waves.

Pickup Angle/Coverage Angle: The effective arc of coverage of a microphone, usually taken to be within the 3dB down points in its directional response.

Pitch: The fundamental or basic frequency of a musical note.

Polar Pattern (Directional Pattern, Polar Response): A graph showing how the sensitivity of a microphone varies with the angle of the sound source, at a particular frequency. Examples of polar patterns are unidirectional and omnidirectional.

Polarization: The charge or voltage on a condenser microphone element.

Pop Filter: An acoustically transparent shield around a microphone cartridge that reduces popping sounds. Often a ball-shaped grille, foam cover or fabric barrier.

Pop: A thump of explosive breath sound produced when a puff of air from the mouth strikes the microphone diaphragm. Occurs most often with “p” and “b” sounds (forward) and “d”, “t”, and “k” sounds (downward).

Presence Peak: An increase in microphone output in the

“presence” frequency range of 2,000 Hz to 10,000 Hz. A presence peak increases clarity, articulation, apparent closeness, and “punch.”

Proximity Effect: The increase in bass occurring with most unidirectional microphones when they are placed close to an instrument or vocalist (within 1 foot). Does not occur with omnidirectional microphones.

Rear Lobe: A region of pickup at the rear of a supercardioid or hypercardioid microphone polar pattern. A bidirectional microphone has a rear lobe equal to its front pickup.

Reflection: The bouncing of sound waves back from an object or surface which is physically larger than the wavelength of the sound.

Refraction: The bending of sound waves by a change in the density of the transmission medium, such as temperature gradients in air due to wind.

Resistance: The opposition to the flow of current in an electrical circuit. It is analogous to the friction of fluid flowing in a pipe.

Reverberation: The reflection of a sound a sufficient number of times that it becomes non-directional and persists for some time after the source has stopped. The amount of reverberation depends on the relative amount of sound reflection and absorption in the room.

Rolloff: A gradual decrease in response below or above some specified frequency.

Sensitivity: A rating given in dBV to express how “hot” the microphone is by exposing the microphone to a specified sound field level (typically either 94 dB SPL or 74 dB SPL). This specification can be confusing because manufacturers designate the sound level different ways. Here is an easy reference guide: 94 dB SPL = 1 Pascal = 10 microbars. To compare a microphone that has been measured at 74 dB SPL with one that has been measured at 94 dB SPL, simply add 20 to the dBV rating. E.g. -40 dBV/Pa = -60 dBV/microbar.

Shaped Response: A frequency response that exhibits significant variation from flat within its range. It is usually designed to enhance the sound for a particular application.

Signal to Noise Ratio: The amount of signal (dBV) above the noise floor when a specified sound pressure level is applied to the microphone (usually 94 dB SPL).

Sound Chain: The series of interconnected audio equipment used for recording or sound reinforcement.

Sound Reinforcement: Amplification of live sound sources.

Speed of Sound: The speed of sound waves, about 1130 feet per second in air.

SPL: Sound Pressure Level is the loudness of sound relative to a reference level of 0.0002 microbars.

Standing Wave: A stationary sound wave that is reinforced by reflection between two parallel surfaces that are spaced a wavelength apart.

Supercardioid Microphone: A unidirectional micro phone with tighter front pickup angle (115 degrees) than a cardioid, but with some rear pickup. Angle of best rejection is 126 degrees from the front of the microphone, that is, 54 degrees from the rear.

Timbre: The characteristic tone of a voice or instrument; a function of harmonics.

Transducer: A device that converts one form of energy to another. A microphone transducer (cartridge) converts acoustical energy (sound) into electrical energy (the audio signal).

Transient Response: The ability of a device to respond to a rapidly changing input.

Unbalanced: A circuit that carries information by means of one signal on a single conductor.

Unidirectional Microphone: A microphone that is most sensitive to sound coming from a single direction-in front of the microphone. Cardioid, supercardioid, and hypercardioid microphones are examples of unidirectional microphones.

Voice Coil: Small coil of wire attached to the diaphragm of a dynamic microphone.

Voltage: The potential difference in an electric circuit. Analogous to the pressure on fluid flowing in a pipe.

Wavelength: The physical distance between the start and end of one cycle of a sound wave.