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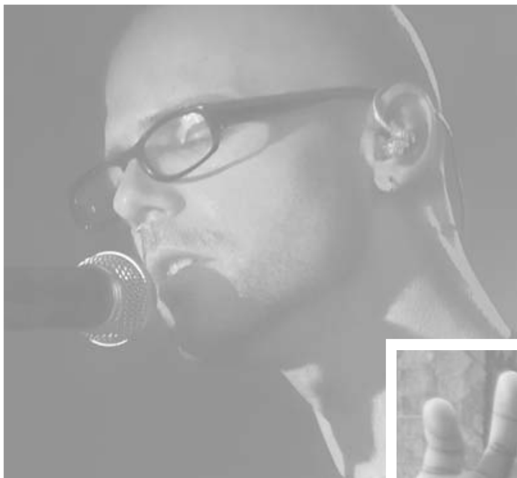
SELECTION AND OPERATION

PERSONAL MONITOR SYSTEMS

THIRD EDITION By Gino Sigismondi



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Introduction

We live in an age of extreme technological advances, and the audio industry is no exception. Average listeners, as well as musicians, have been conditioned by pristine, well-produced CDs and huge, multi-channel surround-sound systems. High quality audio is no longer just the domain of the audiophile and the recording studio engineer, but live concert sound reinforcement as well. Advances in sound system technology treat concert-goers to shows that are bigger, louder, and offer greater clarity than ever before.

The on-stage performer, however, is still dealing with a less-than-perfect solution for on-stage monitoring that was invented and, besides a few small exceptions, exploited to its practical limit in the 1970's. Traditional loudspeaker monitor systems present myriad problems for the performer, as well as the audience, the FOH engineer, and the frazzled monitor engineer. Why should the artists be denied the opportunity to hear their performance in the best possible way?

Personal monitoring is the logical solution. Formerly the province of high-end touring professionals, recent advances in wireless technology and universal earphone development have greatly reduced the initial investment costs for entrance into the world of personal monitors. As with most new technology, though, much misinformation and reluctance to experiment rise to the surface. In an effort to educate and enlighten anyone with an interest in improving the on-stage listening experience, Shure presents "Selection and Operation of Personal Monitor Systems." The booklet gives a short history of monitoring, and describes in detail the benefits of using personal monitors. It also provides specific information on choosing the proper system to meet your needs, and the various ways personal monitor systems can be configured.

The field of personal monitoring is still growing and the technology continues to advance, so let us guide you through the possibilities and the advantages…

CHAPTER ONE

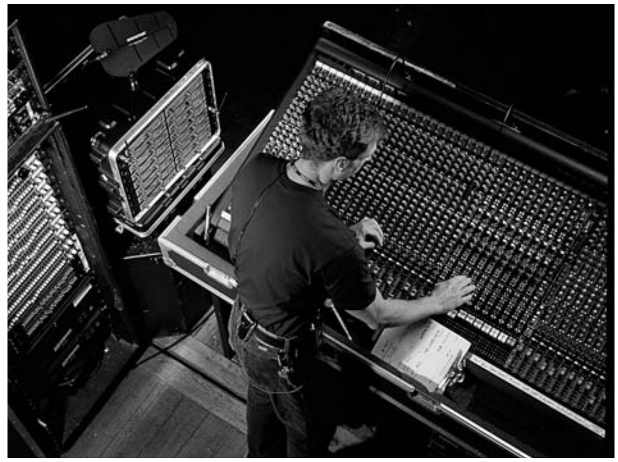
THE (BRIEF) HISTORY OF MONITORS

Radio Wave Transmission

Although no one seems to know for sure, we can probably thank heavy metal bands from the late Sixties/early Seventies for necessitating the development of stage monitoring. Prior to the days of arena concerts and stacks of Marshall™ amplifiers, it wasn't that difficult to hear your voice coming through the main PA loudspeakers. Most concerts were held in smaller venues, with a few notable exceptions. When the Beatles played Shea Stadium in 1964, the only PA was for voice; guitars were only as loud as the guitar amplifiers. Of course, the crowd noise was so loud even the audience couldn't hear what was going on, let alone the band! As rock and roll shows continued to get bigger and louder, it became increasingly difficult for performers to hear what they were doing. The obvious solution was to turn some of the loudspeakers around so they faced the band. A further refinement came in the form of wedge-shaped speakers that could be placed on the floor, facing up at the band. Besides being a convenient place for bass players to put their feet, wedge monitors finally gave singers the ability to hear themselves at a decent volume.

With the size of stages increasing, it became difficult to hear everything, not just the vocals. The drums were on risers 15 feet in the air, and guitar amps were occasionally stowed away under the stage. These changes required the use of a monitor console – a separate mixer used for the sole purpose of creating multiple monitor mixes – to accommodate all the additional inputs as well as create separate mixes for each performer. Today, even the smallest music clubs offer at least two or three separate monitor mixes, and it is not uncommon for local bands to carry their own monitor rig capable of handling four or more mixes. Many national touring acts routinely employ upwards of 16 stereo mixes.

Monitor Wedge



Professional Monitor Console

The problems created by traditional monitor systems are numerous; we'll take a detailed look at them in the next chapter. Suffice it to say, a better way to monitor needed to be found. Drummers have used headphones for years to monitor click tracks (metronomes) and loops. Theoretically, if all performers could wear headphones, the need for monitor wedges would be eliminated. Essentially, headphones were the first personal monitors – a closed system that doesn't affect or depend on the monitoring requirements of the other performers. Unfortunately, they tend to be cumbersome and not very attractive. Recent advances in the miniaturization of transducers have allowed performers to use "earphones," essentially headphones reduced to a size that fit comfortably in the ear. Professional musicians, including Peter Gabriel and the Grateful Dead, were among the first to employ this new technology.

The other major contribution to the development of personal monitors is the growth of wireless microphone systems. Hardwired monitor systems are fine for drummers and keyboardists that stay relatively stationary, but other musicians require greater mobility. Wireless personal monitor systems, essentially wireless microphone systems in reverse, allow the performer complete freedom of movement. The first personal monitor systems were prohibitively expensive; only major touring acts could afford them. As with any new technology, as usage becomes more widespread, prices begin to drop. Current personal monitor systems have progressed to a point where they are within any performer's reach.

CHAPTER TWO

WHY USE PERSONAL MONITORS?

When was the last time you had a great experience with a wedge monitor system? You could hear everything, no feedback, plenty of volume (without being dangerous), and the monitor engineer instantly responded to your every request? If you can't remember, you're not alone. Anyone who has performed live has probably dealt with a poor monitor system, but even a great system has many limitations due to the laws of physics, and those laws bend for no one. The concept of personal monitoring rose from the desire to create an on-stage listening experience that could overcome the limitations imposed by a traditional floor monitor system.

Let's define a personal monitor system. Many parallels exist between personal monitors and a traditional floor wedge setup. The purpose of any monitor system is to allow performers to hear themselves. The sounds to be monitored need to be converted to electronic signals for input to the monitor system. This is usually accomplished via microphones, although in the case of electronic instruments such as keyboards and electronic drums, the signals can be input directly to a mixing console. The various signals are then combined at a mixer, and output to either power amplifiers and loudspeakers or to the inputs of personal monitor systems. Any amount of signal processing, such as equalizers or dynamics processing (compressors, limiters, etc.), can be added in-between. A hardwired personal monitor system is similar (in signal flow terms) to a traditional wedge system, since the belt pack is basically a power amplifier, and the earphones are tiny loudspeakers. A wireless personal monitor system, however, adds a few more components, specifically a transmitter and receiver. From the output of the mixer, the audio signal goes to a transmitter, which converts it to a radio frequency (RF) signal. A belt-pack receiver, worn by the performer, picks up the RF signal and converts it back to an audio signal. At this stage the audio is then amplified and output to the earphones. See Chapter 4 for a complete discussion of the various earphone types available.

So just what is the "experience" of personal monitors? The four most prominent benefits when using them are listed below:

- *Superior sound quality*
- *Portability*
- *Mobility*
- *Personal Control*

Hardwired System

Wireless System

The term "personal monitors" is derived from several factors, but basically revolves around the concept of taking a monitor mix and tailoring it to each performer's specific needs, without affecting the performance or listening conditions of the others. The concept is broader than that of "in-ear monitoring", which states where the monitors are positioned, but gives no further information on the experience.

Superior Sound Quality

There are several factors that, when taken as a whole, result in the superior sound quality of personal monitor systems. These factors include adequate volume for the performers, gain-before-feedback, hearing conservation, reduced vocal strain, and less interference with the audience mix.

Adequate Volume

The most common request given to monitor engineers is "Can you turn me up?" (Sometimes not phrased quite so politely.) Unfortunately, it is not always quite that simple. Many factors can limit how loud a signal can be brought up when using traditional floor monitors: size of the power amplifiers, power handling of the speakers, and most importantly, potential acoustic gain (see Gain-Before-Feedback below). Another factor that makes hearing oneself difficult is the noise level onstage. Many times, vocalists rely solely on stage monitors, unlike guitarists, bassists, and keyboardists whose instruments are generally amplified to begin with. Drummers, of course, are acoustically loud without amplification. Volume wars are not uncommon as musicians struggle to hear themselves over the ever-increasing din. The clarity of the vocals is often obscured as other instruments are added to the monitor mix, which becomes increasingly necessary if fewer mixes are available. Keyboards, acoustic guitars, and other instruments that rely on the monitors often compete with the vocals for sonic space. A personal monitor system, which isolates the user from crushing stage volumes and poor room acoustics, allows the musician to achieve a "studio-like" quality in the onstage listening experience. Professional, isolating earphones, when used properly, provide between 10 and 20 dB of reduction in background noise level (see Chapter 4 for more information on earphones). The monitor mix can then be tailored to individual taste without fighting against otherwise uncontrollable factors.

Gain-Before-Feedback

When it comes to achieving higher monitoring levels with traditional stage wedges, you can always add more amplifiers and more loudspeakers, but you cannot defy the laws of physics. The concept of gain-before-feedback relates to how loud a microphone can be turned up before feedback occurs.

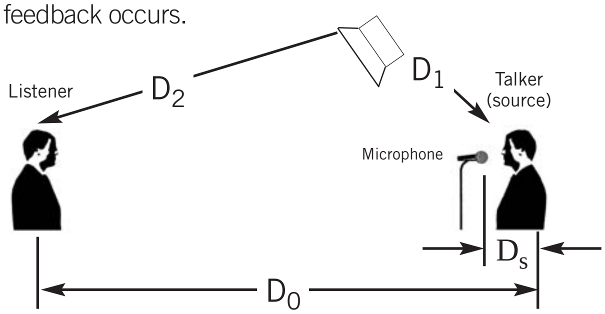


Figure 1: Potential Acoustic Gain

$$PAG = 20 (\log D_1 - \log D_2 + \log D_0 - \log Ds) - 10 \log NOM -6$$

Closely related is PAG, or Potential Acoustic Gain. The PAG equation is a mathematical formula that allows you to predict how much gain is available in your sound system before reaching the feedback threshold, by plugging in known factors such as source-to-microphone distance and microphone-to-loudspeaker distance (see figure 1). Simply stated, the further away you get from the microphone, or the closer the microphone is to the loudspeaker, or the further away the loudspeaker is from the listener, then the less available gain-before-feedback. Now picture a typical stage. The microphone is close to your mouth; that's good. The microphone is close (relatively) to the monitor loudspeaker; that's bad. The monitor loudspeaker is far (relatively) from your ears; that's also bad. Feedback occurs whenever the sound entering a microphone is reproduced by a loudspeaker and "heard" by the same microphone again. To achieve a decent monitoring level, you need quite a bit of available gain. But given the above situation, you have two major factors working against you. Compounding the problem is the issue of NOM, or Number of Open Microphones. Every time you double the number of open microphones, the available gain-before-feedback drops by 3 dB. With four open microphones on stage instead of one, the available gain has dropped by 6 dB. What can you do? The PAG equation assumes omnidirectional microphones, so using cardioid or even supercardioid pattern microphones will help; just don't point them at the speakers. Also, the equation assumes that the sound system has a perfectly flat frequency response. The most commonly employed tool for reducing feedback due to response problems is the graphic equalizer. Since some frequencies will feedback before others, an equalizer allows a skilled user to reduce the monitor system's output of those troublesome frequencies. This technique results in approximately 3-9 dB of additional gain, assuming the microphone position doesn't change. It is common practice for some monitor engineers to attempt to equalize the monitor system to the point where there is no feedback, even with a microphone pointed right into the speaker cone. Unfortunately, the fidelity of the monitor is often completely destroyed in an effort to eliminate feedback using equalizers. Even after equalization has "flattened" the response of the monitor system, PAG again becomes the limiting factor. At this point, you can't get any closer to the microphone, and moving the loudspeaker closer to your ears also makes it closer to the microphone, negating any useful effect on PAG.

Personal monitoring completely removes PAG and gain-before-feedback issues. The "loudspeakers" are now sealed inside your ear canal, isolated from the microphone. With the feedback loop broken, it is possible to achieve as much volume as necessary – which leads to the next topic...

Hearing Conservation

The main purpose of personal monitors is to hear yourself better. But it doesn't do any good if you can't hear at all. As mentioned earlier, volume wars on stage are a universal problem. Prolonged exposure to extremely high sound pressure levels can quickly cause hearing to deteriorate. Some performers have taken to wearing ear plugs to protect their hearing, but even the best ear plugs cause some alteration of frequency response. Personal monitors offer a level of hearing protection equal to that of ear plugs, but with the additional benefit of tiny loudspeakers in the plugs. The monitoring level is now in the hands of the performer. If it seems to be too loud, there is no excuse for not turning the monitors down to a comfortable level. The use of an onboard limiter is strongly recommended to prevent high level transients from causing permanent damage. In larger, complex monitor rigs, outboard compressors and limiters are often employed to offer a greater degree of control and protection.

NOTE: Using a personal monitor system does not guarantee that you will not or can not suffer hearing damage. These systems are capable of producing levels in excess of 130 dB SPL. Prolonged exposure to these kinds of levels can cause hearing damage. It is up to the individual user to be responsible for protecting his or her own hearing. Please see Chapter 8 for more information on safely using personal monitors.

Reduced Vocal Strain

Closely related to the volume issue, the ability to hear more clearly reduces vocal strain for singers. In order to compensate for a monitor system that does not provide adequate vocal reinforcement, many singers will force themselves to sing with more power than is normal or healthy. Anyone who makes a living with their voice knows that once you lose it, you lose your livelihood. Every precaution should be taken to protect your "instrument," and personal monitors are a key ingredient in helping vocalists continue to sing for years to come (See Adequate Volume, previously discussed.)

Stereo

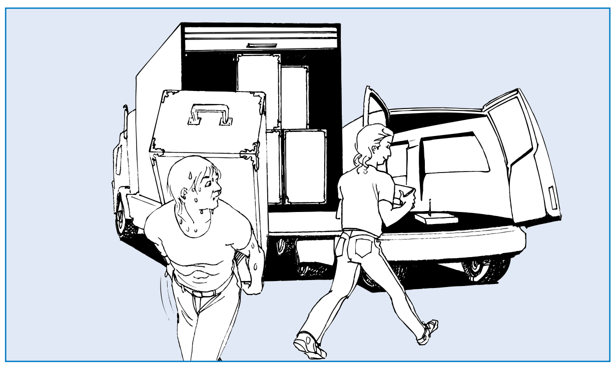
A distinct advantage of using a personal monitor system is the ability to listen in stereo. While it may not be applicable to all situations, especially with a limited number of mixes available, a monitor mix created in stereo can more accurately recreate a realistic listening environment. We spend our entire lives listening in stereo; logically, a stereo monitor mix increases the perception of a natural sound stage. Monitoring in stereo can also allow for lower overall listening levels. Imagine a group with two guitar players sharing the same mix. Both instruments are occupying the same frequency spectrum, and in order for each guitarist to hear, they are constantly requesting their own level be turned up. When monitoring in mono, the brain interprets sounds based only on amplitude and timbre. Therefore, when two sounds have roughly the same timbre, the only clue the brain has for perception is amplitude, or level. Stereo monitoring adds another dimension, localization. If the guitars are panned, even slightly, from center, each sound occupies its own "space." The brain uses these localization cues as part of its perception of the sound. Research has shown that if the signals are spread across the stereo spectrum, the overall level of each signal can be lower, due to the brain's ability to identify sounds based on their location.

Interference with the Audience Mix

The benefits of personal monitors extend beyond those available to the performer. An unfortunate side-effect of wedge monitors is spill from the stage into the audience area. Although directional at high frequencies, speaker cabinets radiate low frequency information in a more or less omnidirectional manner. This situation aggravates the already complex task facing the FOH (front-of-house) engineer, who must fight against loud stage volumes when creating the audience mix. The excessive low frequencies coming off the backs of the monitors make the house mix sound "muddy" and can severely restrict the intelligibility of the vocals, especially in smaller venues. But eliminate the wedges, and the sound clears up considerably.

Portability

Portability is an important consideration for performing groups that travel, and for installations where the sound system or the band performance area is struck after every event. Consider the average monitor system that includes 3 or 4 monitor wedges at roughly 40 pounds each, and one or more power amplifiers at 50 pounds – this would be a relatively small monitor rig. A complete personal monitor system, on the other hand, fits in a briefcase. Purely an aesthetic consideration, removing wedges and bulky speaker cables from the stage improves the overall appearance. This is of particular importance to corporate/wedding bands and church groups, where a professional, unobtrusive presentation is as important as sound quality. Personal monitors result in a clean, professional-looking stage environment.



Personal Monitors Won't Break Your Back.

Mobility

Monitor wedges produce a "sweet spot" on stage; a place where everything sounds pretty good. If you move a foot to the left or right, suddenly things do not sound as good anymore (see figure 2). The relatively directional nature of loudspeakers, especially at high frequencies, is responsible for this effect. Using personal monitors, though, is like using headphones – the sound goes where you go. The consistent nature of personal monitors also translates from venue to venue. When using wedges, room acoustics play a large part in the overall quality of the sound. Since professional earphones form a seal against ambient

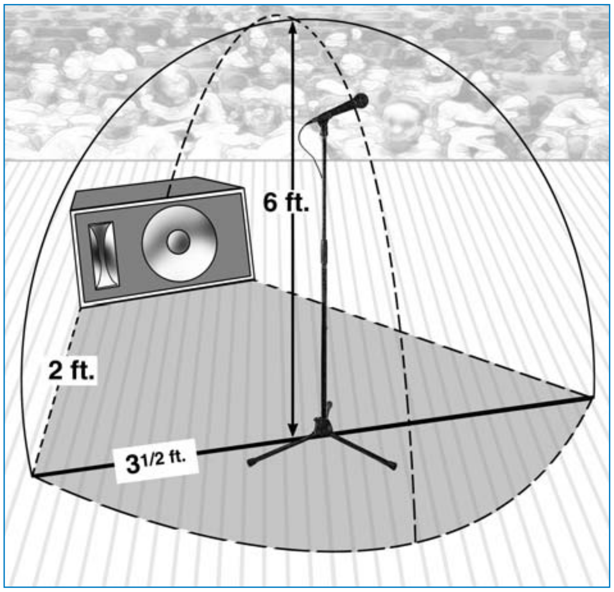


Figure 2: Sweet Spot Created by a Monitor Wedge

noise, acoustics are removed from the equation. In theory, given the same band with the same members, the monitor settings could remain virtually unchanged, and the mix will sound the same every night.

Personal Control

Perhaps the most practical benefit to personal monitors is the ability to have direct control over what you are hearing. While still relying on the sound engineer to make fine adjustments, personal monitor systems give the performer some ability to make broad adjustments, such as overall volume, pan, or the ability to choose different mixes. If everything in the mix needs to be louder, instead of giving a series of complex hand gestures to the monitor engineer, the performer can raise the overall volume directly from the belt-pack. Further personal control is provided by systems that feature a "dual-mono" mode, where the belt-pack combines the left and right audio channels of a stereo system and sends the combined signal to both sides of the earphones (see figure 3). The inputs to the system should now be treated as "Mix 1" and "Mix 2" instead of left and right. The balance control on the receiver acts as a mix control, allowing the performer to choose between two mixes, or listen to a combination of both mixes with control over the level of each. Panning to the left gradually increases the level of "Mix 1" in both ears, while reducing the level of "Mix 2," and vice versa. Chapter 5 includes some practical applications for dual-mono monitoring. Putting a small, outboard mixer near the performer increases the amount of control. By giving control of the monitor mix to the performer, the sound engineer can spend more time concentrating on making the band sound good for the audience instead of worrying about making the band happy.

Lesser expensive, mono-only systems can offer a similar type of control by providing multiple inputs at the transmitter, with a separate volume control for each. Consequently, the transmitter should be located near the performer for quick mix adjustments.

The cost of transitioning to personal monitors has recently dropped dramatically. A basic system costs as much, if not less than, a typical monitor wedge, power amplifier, and graphic equalizer combination. Expanding a system is also more cost effective. When providing additional wedges for reproducing the same mix, a limited number can be added before the load on the amplifier is too great, and another amp is required. With a wireless personal monitor system, however, the number of receivers monitoring that same mix is unlimited. Additional receivers do not "load" the transmitter, so feel free to add as many receivers as necessary without adding more transmitters. For bands that haul their own PA, transportation costs may be reduced as well. Less gear means a smaller truck, and possibly one less roadie.

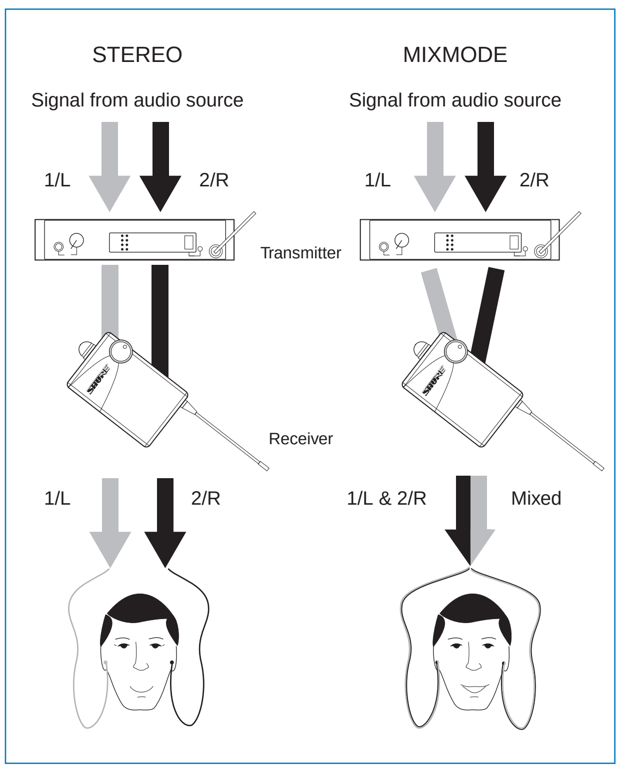


Figure 3: Graphic Representation of Stereo vs. Dual Mono

CHAPTER THREE

CHOOSING A SYSTEM

Given the nature of personal monitoring, choosing the right system is an important step. Several choices are available. Present as well as future needs should be taken into account before making an investment.

Wireless vs. Hardwired

Personal monitor systems come in two basic varieties – wireless or hardwired. A hardwired system requires the performer be tethered to a cable, which is not necessarily a negative. Drummers and keyboard players who stay relatively stationary, or even back-up singers, can take advantage of the lower cost and greater simplicity of a hardwired personal monitor system. Simply connect the monitor sends to the inputs of the hardwired beltpack and dial up a mix. Hardwired systems also work worldwide without the hassle of finding clear frequencies or dealing with local wireless codes. Lastly, if several performers require the same mix, hardwired systems can be daisy-chained together without significant signal loss, if the input impedance is sufficiently high to allow multiple systems to be connected to a single output with Y-cables. Another way to connect more than one hardwired system to the same output is to use a distribution amplifier, a device that takes a single input and splits it to multiple outputs, each with its own individual level control.

Wireless equipment, by nature, requires special considerations and attention to detail (see Chapter 7: Wireless Basics). But the advantages many times outweigh the increased cost and complexity. One of the main benefits of personal monitors is a consistent mix no matter where you stand; going wireless allows you to exploit this advantage to its fullest extent. (The Rolling Stones use personal monitors on a remote stage in the middle of the seating area!) Additionally, when several performers require the same mix, hooking them up is even easier. As many wireless receivers as necessary can monitor the same mix with no adverse effects. And of course, no cables to trip on!

Configuring a Personal Monitor System

Choosing the proper system requires a little advance planning to determine the monitoring requirements of your situation.

At the very least, you should know:

1. *How many mixes your situation requires,*
2. *Whether you want to monitor in stereo or mono,*
3. *How many monitor mixes can be created by the mixing consoles you will be using.*

This information directly relates to the equipment that you will need to satisfy the personal monitoring requirements of your band. The following example details the thought process involved in deciding how to configure your system.

1. How many mixes do I need?

The answer to this question depends on how many people are in your band, and their ability to agree upon what they want to hear in the monitors. For example, a typical rock band instrumentation is drums, bass, guitar, keys, lead vocal, and two back up vocals provided by the guitar player and keyboardist. In a perfect world, everyone would want to listen to the same mix, so the answer to this question would be one mix. Of course this would defeat one of the main benefits of "personal" monitors, so let's assume this isn't the case (which isn't too much of a stretch…). An inexpensive configuration uses two mixes, one consisting of vocals, the other of instruments. Using a system that features dual-mono operation, the performers individually choose how much of each mix they wish to hear (see figure 5). This scenario is a cost-effective way to get into personal monitors, yet still requires a fairly good degree of cooperation among band members. Another scenario gives the drummer a separate mix (see figure 6). This option works well for two reasons: 1), drummers, in general, will want to hear considerably more drums in the monitors than other band members, and 2), for bands who perform on small stages the drums are so loud that they are easily heard acoustically (with no additional sound reinforcement). Therefore, drums may not even be necessary in the other mixes. So now we are up to three – the vocal mix, the instruments (minus drums), and the drummer's mix.

Up to this point, we have assumed that the vocalists are able to agree on a mix of the vocal microphones. While forcing singers to share the same mix encourages a good vocal "blend", this theory commonly falls apart in practice (due to a phenomenon we'll call the "more me" syndrome). Often, separating out the lead vocalist to an individual mix will address this issue, and this can be handled in one of two ways. First, place some of the backup vocal mics in the "instruments" mix, and adjust the "vocal" mix to satisfy the lead singer, even if that means adding some instruments to the "vocal" mix. We now have:

- an individual mix for the lead singer,
- a mix for the guitarist and keyboardist that includes their vocals, and
- a drum mix (at this point the bass player can drop in wherever he/she wants, often on the drummer's mix), yet we are still only using three mixes.

The second option is to create a fourth mix for the lead singer, without affecting the other three. This configuration allows the guitarist and keyboardist to retain control between their vocals and instruments, while giving the lead singer a completely customized mix. Does the bass player need a separate mix? Now you are up to five mixes. Adding a horn section? That could easily be a sixth mix. So where does it end? Well, several other factors (including your budget!) will help decide at which point you limit the number of mixes.

2. Do I want to monitor in stereo or mono?

Most personal monitor systems allow for monitoring in either stereo or mono. At first glance, stereo may seem the obvious choice, since we hear in stereo, and everything else these days features stereo sound - CDs, TV, VCRs, even your computer. Stereo, by its very nature, requires two channels of audio. What this means for personal monitor users is two sends from the mixer to create a stereo monitor mix – twice as many as it takes to do a mono mix (see figure 7). Stereo monitoring can rapidly devour auxiliary sends; if your mixer only has four sends, you can only create two stereo mixes, versus four mono.

While not quite as “realistic” as stereo monitoring, mono allows more mixes from a smaller mixing console, and sometimes fewer transmitters. Some stereo transmitters can be operated in a “dual-mono” mode, which provides two mono mixes instead of one stereo. This capability can be a great way to save money. If you only need one mono mix, or if you are a solo artist or jobbing musician looking to have your own personal system, mono-only systems are another cost-effective option. Strongly consider a system that includes a microphone input that will allow you to connect your primary instrument directly to the monitor system.

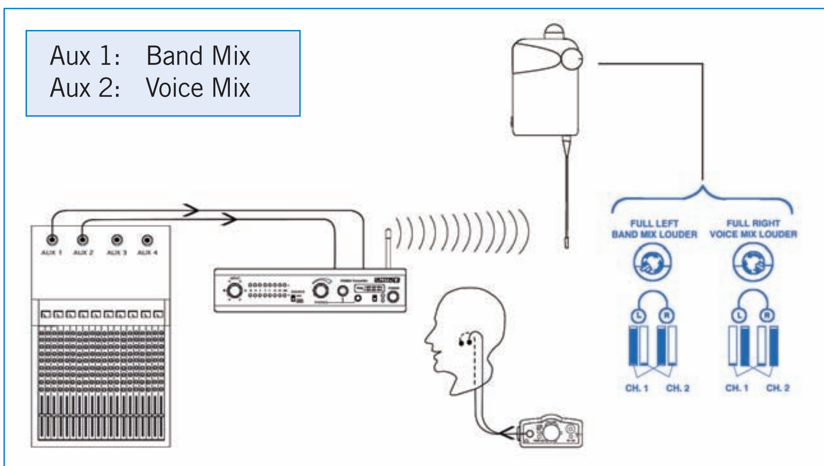


Figure 5: Two Mixes Using Dual Mono

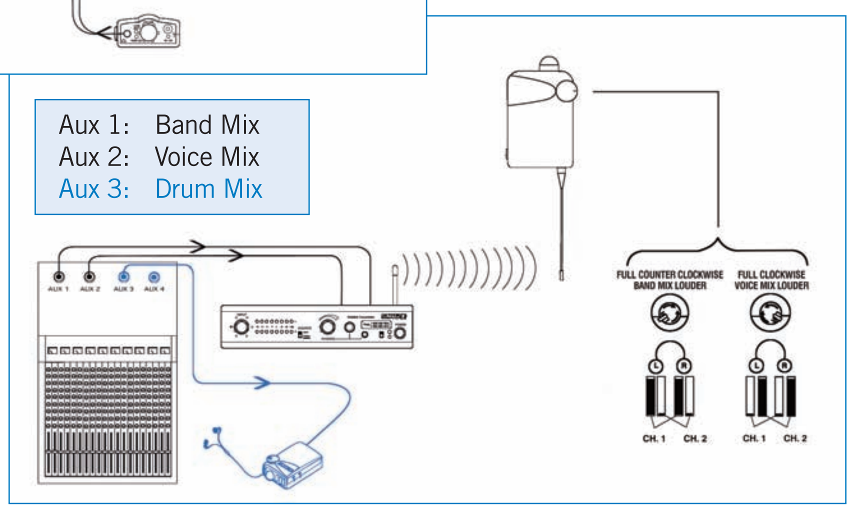


Figure 6: Three Mixes Using Dual Mono

3. How many mixes are available from the front of house console?

Monitor mixes are typically created using auxiliary (Aux) sends from a mixer, either the front-of-house (audience) console, or a dedicated monitor console if it's available. A typical small-format console will have at least four auxiliary sends. Whether or not all these are all available for monitors is another matter. Aux sends are also used for effects (reverb, delay, etc.). At any rate, available auxiliary sends are the final determinate for the number of possible monitor mixes. If your answer to question 1 is greater than the answer to question number 3, you have two options: reconfigure your monitor mixes to accommodate the mixer, or get a new mixer. Keep in mind that if you don't have a mixer and rely on the equipment supplied by the venues where you play, be prepared to deal with the number of monitor mixes they are equipped to provide.

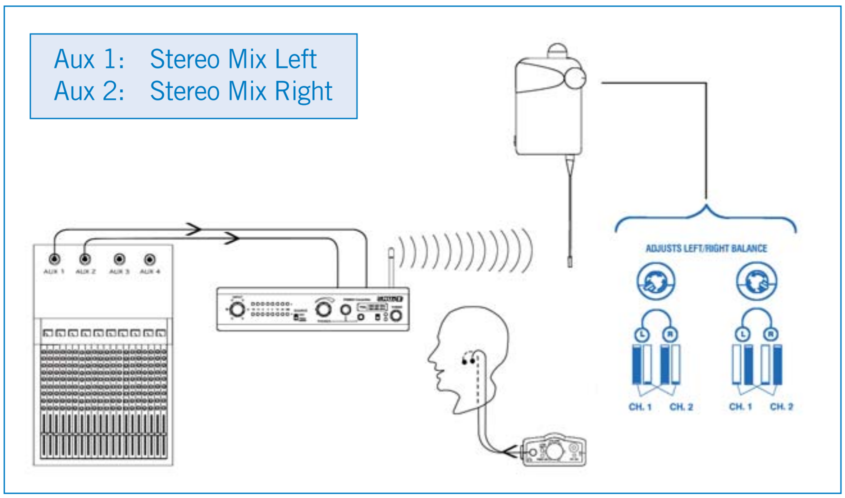


Figure 7: One Stereo Mix

4. How many components will I need?

After you have answered the above questions, plug the numbers into the following equations to determine exactly how many of each component you will need, and choose a system that can handle these requirements.

Stereo:

Number of transmitters = number of desired mixes

Number of required aux sends = 2 (number of transmitters)

(e.g. 4 mixes requires 4 transmitters and 8 aux sends)

Dual Mono:

Number of transmitters = number of desired mixes/2

Number of required aux sends = 2 (number of transmitters)

(e.g. 4 mixes requires 2 transmitters and 4 aux sends)

Mono only:

Number of transmitters = number of desired mixes

Number of aux sends = number of transmitters

(e.g. 4 mixes requires 4 transmitters and 4 aux sends)

Number of receivers = number of performers

CHAPTER FOUR

EARPHONES

Earphone Options

The key to successful personal monitoring lies in the quality of the earphone. All the premium components in the monitoring signal path will be rendered ineffective by a low quality earphone. A good earphone must combine full range audio fidelity with good isolation, comfort, and inconspicuous appearance. The types of earphones available include inexpensive iPod®-type "ear-buds", custom molded earphones, and universal earphones. Each type has its advantages and disadvantages. While relatively affordable, "ear-buds" have the poorest isolation (see Chapter 8), and are not really designed to withstand the rigors of a working musician's environment (not to mention they don't stay in your ears when you're headbanging!). On the other end of the spectrum, custom molded earphones (see figure 8) offer exceptional sound quality and isolation, a considerably higher price tag, and are difficult to try before buying since they are made specifically for one person's ears. The procedure for getting custom molds involves a visit to an audiologist. The audiologist makes an impression of your ear canals by placing a "dam" inside your ear to protect your ear drum, and fills them with a silicone based material that conforms exactly to the dimensions of your ears. The impressions are then used to create the custom molded earphones. Another visit to the audiologist is required for a final fitting.

Universal earphones combine the superior sound quality and isolation of the custom molded designs with the versatility and "out-of-the-box" readiness of the consumer phones. The "universal" nature is attributed to the interchangeable sleeves that are typically used to adapt the earphone to the performer's ear canal. This design allows the user to audition the various sleeves to see which works best for them, as well as being able to demo the earphones before a purchase is made. Different ear piece sleeves include foam, flexible rubber sleeves, rubber flange tips, and custom molded, each with their own advantages and disadvantages.

Figure 8: Custom Molded Earphones
(ProPhonic 2X-S Courtesy Sensaphonics)

Figure 9: Single Transducer Earphones

Foam

The most common sleeve option, foam sleeves resemble regular foam ear plugs, but with a small hole in the center of the foam lined with a tube of plastic (see figure 9). The shaft of the earphone is inserted through this hole. To properly use the earphone, compress the foam as small as possible, and insert it into your ear canal. Hold it in place until the foam expands completely, forming a tight seal. Drape the cable over the top of, and behind, your ear. Repeat this procedure for your other ear. If the earphone cable has a plastic adjustment tube, use it to reduce excess slack. Foam sleeves offer excellent isolation and good low frequency performance. On the downside, they eventually get dirty and worn, and need to be replaced. Proper insertion of the foams also takes longer – relative to the other options – since you need to hold the earphone in place while the foam expands.

Rubber Sleeves

For quick insertion and removal of the earphones, flexible rubber sleeves may be a good choice (see figure 10). Made of soft, flexible plastic, they resemble a mush-room cap and are usually available in multiple sizes to accommodate various ear canal shapes. Insertion is very easy, just place the sleeve in your ear canal and drape the cable as detailed above. While the seal is usually not as tight as with the foams, they are washable and reusable.

Figure 10: Earphone Sleeves

Triple-flange Sleeves

A variation of the standard rubber sleeves, the triple-flange sleeves have three rings (or flanges) around a central rubber tube (see figure 11). They are sometimes referred to as "Christmas trees" based on their shape. The pros and cons are similar to that of the flex sleeves, but with better isolation and a different comfort factor that some users may find more to their liking.

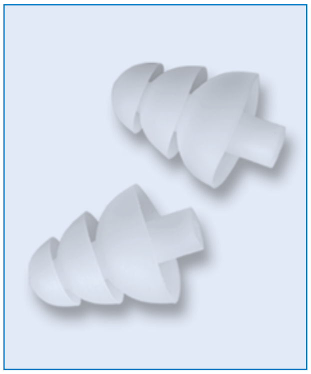


Figure 11: Triple-Flange Sleeves

Custom Sleeves

The fourth, and most expensive, option is custom molded sleeves. The custom sleeves combine the relative ease of insertion and permanency of the rubber sleeves with the superior (depending on the preference of the user) isolation of the foams. The process for obtaining custom sleeves for universal earphones is very similar to that of getting custom molded earphones; a visit to an audiologist is required to get impressions made of your ear canals. Custom sleeves also give the user many of the same benefits as custom molded earphones, but usually at a lower cost, and with the added benefit of being able to interchange earphones with the sleeves if they get lost, stolen, or are in need of repair.

IMPORTANT NOTE: There are several brands of custom molded earplugs with internal filters that have relatively flat frequency response and different levels of attenuation. Although it may be physically possible to make universal earphones fit into the plugs with the filter removed, we strongly recommend against it. The location of the earphone shaft in the ear canal is crucial to obtaining proper frequency response, and most earplugs will prevent them from getting in the proper position. Once again, custom molded earplugs are NOT an acceptable alternative to custom sleeves.

Lastly, if there is ever a problem with a universal earphone, another set can be substituted with no negative repercussions. A custom molded earphone does not allow for this kind of versatility; if one needs repair, the only alternative is to have a back-up (at roughly $500 - $750 a pair!) to use in the interim.

Earphone Transducers

The internal workings of earphones vary as well. There are two basic types of transducer used in earphone design – dynamic and balanced armature.

The dynamic types (see figure 12) work on the same principle as dynamic microphones or most common loudspeakers. A thin diaphragm is attached to a coil of wire suspended in a magnetic field. Diaphragm materials include Mylar (in the case of dynamic microphones) or paper (for loudspeakers). As current is applied to the coil, which is suspended in a permanent magnetic field, it vibrates in sympathy with the variations in voltage. The coil then forces the diaphragm to vibrate, which disturbs the surrounding air molecules, causing the variations in air pressure we interpret as sound. The presence of the magnet-voice coil assembly dictates a physically larger earphone. Dynamic transducers are commonly used in the "ear-bud" types, but recent technological advances have allowed them to be implemented in universal designs. They are also found in some custom molded earphones.

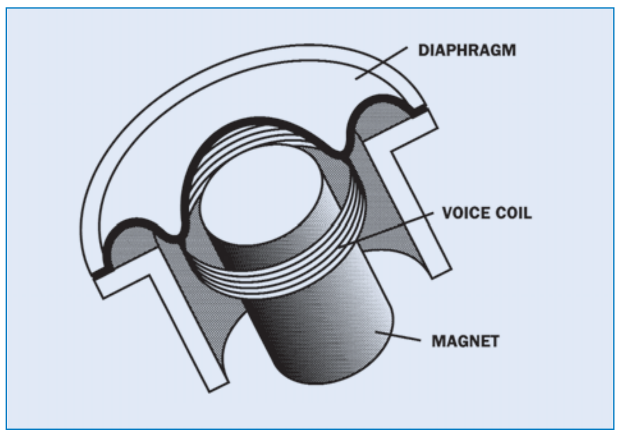


Figure 12: Dynamic Transducer

Originally implemented in the hearing aid industry, the balanced armature transducer (see figure 13) combines smaller size with higher sensitivity. A horseshoe shaped metal "arm" has a coil wrapped around one end and the other suspended between the north and south poles of a magnet. When alternating current is applied to the coil, the opposite arm (the one suspended in the magnetic field) is drawn towards either pole of the magnet. The vibrations are then transferred to the diaphragm, usually a thin layer of foil. Balanced armature transducers are similar to the elements used in controlled magnetic microphones. In addition to the increased sensitivity, they typically offer better high frequency response. Achieving a good seal between the earphone and the ear canal is crucial to obtaining proper frequency response.

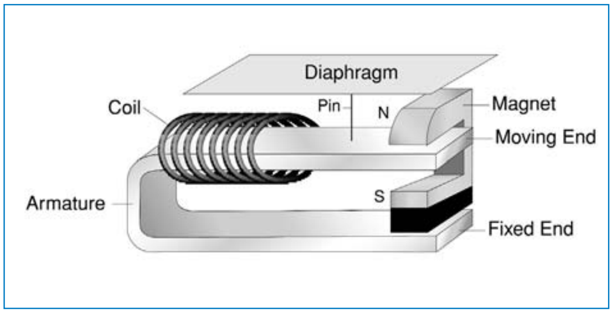


Figure 13: Balanced Armature Transducer

A further subdivision occurs with the use of multiple transducers. Dual transducer (dual driver) earphones are the most common (see figure 14). Another example of dual-transducer design is a two-way loudspeaker with a horn (or tweeter) for high-frequency reproduction and a woofer for low-frequency sounds. The frequency spectrum is divided in two by a crossover network. Each driver only has to reproduce the frequency range for which it has been optimized. Dual driver earphones work on a similar principle – each earphone contains a "tweeter" and a "woofer" optimized for high and low frequency performance, respectively. Additionally, a passive crossover is built into the cable to divide the audio signal into multiple frequency bands. The end result is usually much better low end, as well as extended high frequency response. This trait may be of particular interest to bassists and drummers, whose instruments produce a good deal of low frequency content.

Figure 14: Dual Transducer Earphones

The Occluded Ear

A note for users who are new to earphones: When your ear canal is acoustically sealed ("occluded"), the auditory experience is different from what you may be traditionally accustomed to. For those performers who have spent many years using traditional floor monitors, an adjustment period may be necessary. A common side effect for vocalists is under-singing. The sudden shock of hearing yourself without straining causes some vocalists to sing softer than they normally would, making it difficult for the FOH engineer to get the vocals loud enough in the house mix. Remember, the FOH engineer is still fighting the laws of PAG, so singers still needs to project.

Another side effect of the occluded ear is a buildup of low frequencies in the ear canal. Sealing off the ear canal such as with an earplug, causes the bones of the inner ear to resonate due to sound pressure levels building up in the back of the mouth. This resonance usually occurs below 500 Hz and results in a "hollow" sound that may affect vocalists and horn players. Recent studies have shown, however, that ear molds which penetrate deeper into the ear canal (beyond the second bend) actually reduce the occlusion effect. The deeper seal reduces vibration of the bony areas of the ear canal.

Ambient Earphones

Some users of isolating earphones complain of feeling "closed-off" or too isolated from the audience or performance environment. While isolating earphones provide the best solution in terms of hearing protection, many performers would appreciate the ability to recover some natural ambience. There are several ways in which this can be accomplished, the most common being ambient microphones (see Chapter 6). Ambient microphones are typically placed in a fixed location, nowhere near the listener's ear, and the levels are controlled by the sound engineer instead of the performer. Additionally, the directional cues provided by ambient microphones (assuming a left/right stereo pair) are dependent on the performer facing the audience. If the performer turns around, the ambient cues will be reversed.

More natural results can be obtained by using a newer technology known as ambient earphones. An ambient earphone allows the performer, by either acoustic or electronic means, to add acoustic ambience to the personal monitor mix. Passive ambient earphones have a port, essentially a hole in the ear mold, that allows ambient sound to enter the ear canal. While simple to implement, this method offers little in the way of control and could potentially expose the user to dangerous sound pressure levels. Active ambient earphones use miniscule condenser microphones mounted directly to the earphones. The microphones connect to a secondary device that provides the user with a control to blend the desired amount of ambience into the personal monitor mix. Since these microphones are located right at the ear, directional cues remain constant and natural. Ambient earphones not only provide a more realistic listening experience, but also ease between-song communication amongst performers.

CHAPTER FIVE

SETTING UP A PERSONAL MONITOR SYSTEM

Hooking up personal monitor systems and making them work is a relatively simple process, but the ways in which they can be configured are almost limitless. In this section, we'll take a look at several typical system set-up scenarios. Personal monitor systems are equally useful for performance and rehearsal, and their benefits extend from small nightclub settings to large arena tours to houses of worship.

Rehearsals

If you already own a mixer, implementing a personal monitor system for rehearsals is a simple process. There are a number of ways you can get signal into the system, depending on how many mixes are necessary. To create a simple stereo mix, simply connect the main outputs of the mixer directly to the system inputs. Auxiliary sends can also be used if separate mixes are desired. For bands that carry their own PA system (or at least their own mixer), this method allows them to create a monitor mix during rehearsal, and duplicate it during a performance. No adjustment needs to be made for the acoustic properties of the performance environment. A monitor mixer with built-in microphone splits is desirable in this application, so microphones can be shared between the stage and front-of-house. When it comes to performance time, your monitor mix can be dialed-in exactly as it was in rehearsal, without affecting the house mix.

Performance

Club/Corporate/Wedding Bands – No Monitor Mixer

The majority of performing groups do not have the benefit of a dedicated monitor mixing board. In this situation, monitor mixes are created using the auxiliary sends of the main mixing console. The number of available mixes is limited by the capabilities of the mixer (see the AuxPander – Appendix B). Most personal monitor systems have at least two inputs. Therefore, select a mixer capable of providing at least two dedicated, prefader auxiliary sends. Prefader sends are unaffected by changes made to the main fader mix. Postfader sends change level based on the positions of the channel faders. They are usually used for effects. Although postfader sends can be used for monitors, it can be somewhat distracting for the performers to hear fader moves. Most users that only have two auxiliary sends available should choose either a stereo system with dual-mono operation or a mono system with at least two inputs, since this allows for the most flexibility. Hookup is easy – just connect Aux Send 1 of your console to the left (or first) input of the transmitter (or hardwired system) and Aux Send 2 to the right (or second) input. (Use Aux 3 and 4 if those are the prefader sends – all consoles are different!) Then, depending on who is listening to what, create the mixes by turning up the auxiliary sends on the desired channels. A few common two-mix setups are listed at the top of this page:

Two Monitor Mixes (Dual Mono)

Option 1	
Aux 1 (Left Input)	Aux 2 (Right Input)
Vocal Mix	Band mix (Guitars, drums, etc.)
Option 2	
Aux 1 (Left Input)	Aux 2 (Right Input)
"Front" mix (Vocals, guitars, horns, etc.)	"Backline" mix (Drums, bass)
Option 3	
Aux 1 (Left Input)	Aux 2 (Right Input)
"Ego" mix (Lead instrument or vocal)	Everything else

Typically, the balance control is used to select the mix the performer wants to hear. Be sure the receivers are set for dual-mono operation, or each mix will only be heard on the left or right side, but not in both ears. Also remember that any number of receivers can monitor the same transmitter.

Some performers may prefer to listen to the house mix, so they can monitor exactly what the audience is hearing. For systems that feature loop-through jacks, rather than try to duplicate the house mix with auxiliary sends, the main outputs of the console can be connected directly to the wireless transmitter. Loop jacks allow the signal to pass on to the main PA, unaffected by the monitor system. Keep in mind that this will not always produce the desired results for the performer. Rarely will what sounds good in the ear canal sound equally as good through a PA system in a less-than-perfect acoustic environment. Many times, a vocal that seems to blend just right for a personal monitor mix will get completely lost through the PA, especially in a small room when live instruments are used. This technique may be appropriate for electronic bands, where the majority of instruments are input directly to the mixer. The only sound in the room is that created by the sound system. The more auxiliary sends a console has, the more monitor mixes you can create. See the tables below for more examples.

Club Level Bands – With Monitor Console

At this point, we've reached the limit of monitoring capabilities for the typical small format FOH console. Bands that have graduated to the next level of performance (larger, more prestigious clubs and theaters or small tours) may find themselves in a position to take advantage of a dedicated monitor console. Most monitor boards are capable of providing at least eight mono (or four stereo) mixes. It now becomes practical for each band member to have his or her own dedicated mix. System hookup is again very simple – the various mix outputs from the monitor console are connected directly to the personal monitor system. Stereo monitoring is a much more viable option due to the large number of mixes available, as well as the presence of a skilled monitor engineer (hopefully!) to tweak the mixes to the point of perfection. Some performers even carry their own monitor console. Due to the consistent nature of personal monitors, a band with the same instrumentation and performers for every show can leave the monitor mix dialed-in on their console. Since venue acoustics can be completely disregarded, a few minor adjustments are all that is typically necessary during sound check. A personal monitor mixer (preferably with split outputs) can also be used to augment the monitor console, if the performer desires some personal control over what they hear. In the past, drummers or keyboard players would use a small mixer and Y-cables to submix their instruments for personal monitors, mixers with split outputs eliminate the need for Y-cables.

A NOTE ON MONITOR MIXING: Performers now have an unprecedented level of personal control over what they are hearing. The temptation to make yourself the loudest thing in the mix is great, but this may not be the best for the situation. Proper blending with the other members of your ensemble will be next to impossible if the mix is skewed too far from reality. Consider big bands that normally play acoustically, or a vocal choir. These types of ensembles create their blend by listening to each other, not just themselves. If the lead trumpet player uses a personal monitor system, and cranks the trumpet up three times louder than everything else, there is no accurate feedback for the musician on whether they are playing too loud or too soft. Remember, great bands "mix" themselves – don't rely entirely on the sound tech to get it right.

Three Monitor Mixes (Dual Mono)

Option 1			
Aux 1 Out (System 1 Left)	Aux 2 Out (System 1 Right)	Aux 3 Out (System 2 Left)	(System 2 Right)
Vocal mix	Band mix	Dedicated drum mix	Unused
Option 2			
Aux 1 Out (System 1 Left)	Aux 2 Out (System 1 Right)	Aux 3 Out (System 2 Left)	(System 2 Right)
Lead vocal	Everything else	Dedicated drum mix	Unused
Option 3			
Aux 1 Out (System 1 Left)	Aux 2 Out (System 1 Right)	Aux 3 Out (System 2 Left)	(System 2 Right)
"Front" mix	"Backline" mix	"Ego" mix (bandleader gets whatever he/she wants!)	Unused

Four Monitor Mixes (Dual Mono – using only 3 Aux Sends and Monitor Loop Jacks)

Option 1			
Aux 1 Out (System 1 Left)	Aux 2 Out (System 1 Right)	Aux 3 Out (System 2 Left)	(System 2 Right)
Vocal mix	Band mix	Horn mix	Band mix (looped from System 1 Right Loop Out Jack)

Four Monitor Mixes (Dual Mono)

Option 1			
Aux 1 Out (System 1 Left)	Aux 2 Out (System 1 Right)	Aux 3 Out (System 2 Left)	Aux 4 Out (System 2 Right)
Lead Vocalist's mix	Guitarist's mix	Bassist's mix	Drummer's mix
Option 2			
Aux 1 Out (System 1 Left)	Aux 2 Out (System 1 Right)	Aux 3 Out (System 2 Left)	Aux 4 Out (System 2 Right)
Vocal mix	Band mix	Horn mix	Vocal/Band mix
Option 3			
Aux 1 Out (System 1 Left)	Aux 2 Out (System 1 Right)	Aux 3 Out (System 2 Left)	Aux 4 Out (System 2 Right)
"Ego" mix (lead vocal/instrument only)	"Ego" mix (everything else)	Band mix	Dedicated Drum mix

Professional Touring System

When budget is no longer a consideration, personal monitoring can be exploited to its fullest capabilities. Many systems used by professional artists on large scale tours often employ greater than 16 stereo mixes. A completely separate, totally personalized mix is provided for every performer on stage. Large frame monitor consoles are a requirement. For example, to provide 16 stereo mixes requires a monitor console with 32 outputs. Effects processing is generally employed to a much larger extent than with a smaller system. When operating a large number of wireless personal monitor systems, RF related issues become much more important (Chapter 7). Frequency coordination must be done carefully to avoid interaction between systems as well as outside interference. Depending on the extent of the touring, a frequency agile system is desirable, if not required. Proper antenna combining, to reduce the number of transmitter antennas in close proximity, is a necessity. Directional antennas may also be used to increase range and reduce the chances of drop-outs due to multipath interference.

Creating a basic monitor mix: One of the advantages of having a professional sound engineer or monitor engineer is years of experience in mixing sound. This skill cannot be learned overnight. For bands that are new to personal monitors, there is strong temptation to try to create a CD quality mix for personal monitors. While this is certainly possible with a trained sound engineer and the right equipment, it is unlikely that someone unfamiliar with the basic concepts behind mixing will be able to successfully imitate a professional mix. Does this mean you shouldn't use personal monitors without a monitor tech? Certainly not! A common mistake made by novices is to put everything but the kitchen sink into the mix. Here's an alternative to the "everything-in-the-mix" method:

1. Put the monitors in your ears, and turn the system on. DO NOT put any instruments in your mix yet!
2. Try to play a song. While you are playing, determine what you need to hear more of.
3. Begin bringing instruments into the mix, one at time. Chances are, you will need vocals first, since those are often the only unamplified "instruments" on stage.
4. Only turn things up as loud as necessary, and resist the temptation to add instruments to the mix that you can hear acoustically.

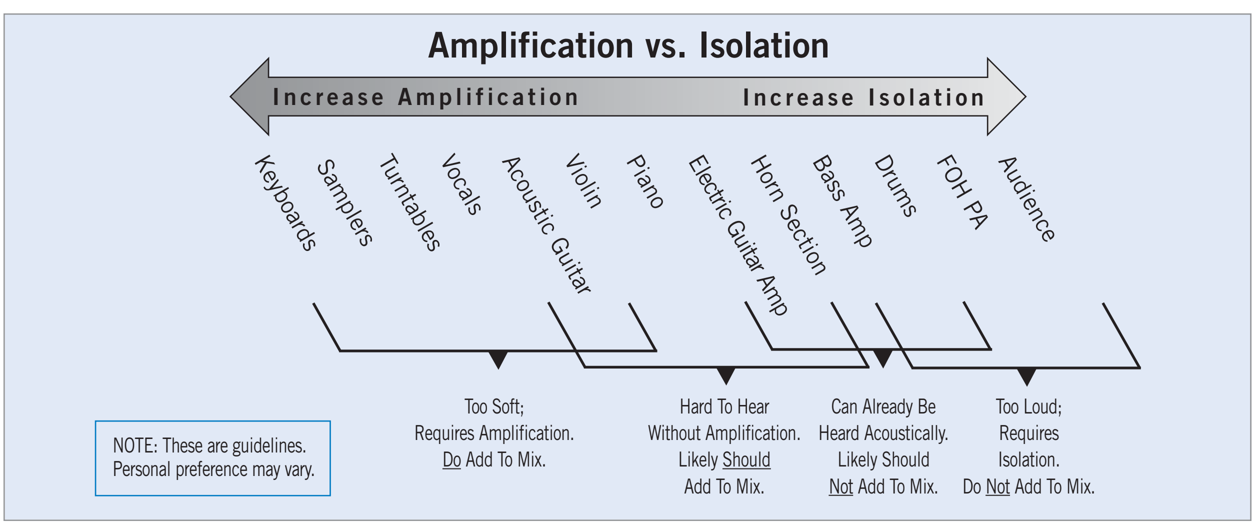
Personal Monitors for Houses of Worship and Sound Contractors

The advantages of using personal monitors extend beyond those of just the performers. We have seen how they benefit the performer, and up to this point we have been discussing personal monitors from a strictly music industry oriented point of view. This section will discuss how personal monitors can be a useful tool for the sound contractor, specifically as they apply to modern houses of worship. Musical performances are rapidly becoming a more prominent part of the worship service. Praise teams and contemporary music groups, while bringing new levels of excitement to traditional church services, also bring with them the problems of your average rock band. Most prominent among these problems are volume wars. Drums naturally tend to be the loudest thing on stage. The guitarist, in order to hear himself better, turns his amplifier up louder. The singers then need more monitor level to compete with the rest of the band. And then the cycle begins again. In any live sound situation, church or otherwise, loud stage volumes can distract from the overall sound quality in the audience. Try an easy experiment at the next sound check. When the band is satisfied with the monitor mix (such as it is…), turn off the audience PA and just listen to the sound coming off the stage. It's probably loud enough that you don't even need to turn on the main system! To compound matters, the "backwash" off the floor monitors consists primarily of low frequency information that muddies-up the audience mix. While this situation creates headaches for most sound engineers, it is even worse in the church environment. The majority of Sunday morning service attendees are not looking for an extremely loud rock and roll concert, but in some

cases the congregation mix gets this loud just so it can be heard over the stage monitors. The naturally long reverberation times of most worship spaces only aggravates the situation. If you turn off the main system and it's still too loud, what can you do? If you turn down the floor monitors, the band complains – not to mention how terrible it will sound.

With the band using personal monitors, these problems evaporate. Traditional floor monitors can be completely eliminated. For part two of our experiment, turn off the stage monitors while the band is playing. Notice how much clearer the audience mix becomes? This is how it would sound if the band were using personal monitors. Also, personal monitors are not just for vocalists. Drummers with personal monitors tend to play quieter. When the loudest instrument on stage gets quieter, everything else can follow suit. Some churches take this a step further by using electronic drums which create little, if any, acoustic noise. Bass, keyboard, and electric guitar can also be taken directly into the mixer if the players are using personal monitors, eliminating the need for on-stage amplifiers. The end result is a cleaner, more controlled congregation mix, and musicians can have loud, clear monitors without affecting the congregation. Secondly, consider the feedback issue. Feedback occurs when the sound created at the microphone comes out of a loudspeaker, and reenters the microphone. The closer the loudspeaker is to the microphone, the greater the chance for feedback. By eliminating the floor monitor, you also eliminate the worst possible (and most likely) feedback loop. With the "loudspeakers" sealed inside the ear canal, there is no chance for the signal to reenter the microphone. No equalizer or feedback reducer will ever be as effective as personal monitors will at eliminating feedback on the stage.

Many other uses are possible for personal monitors. Choir directors could use them for cues, or to hear the pastor more clearly. The isolation provided by personal monitors can be of great benefit to organists, especially when they are located at the opposite end of the sanctuary from the choir. Timing can be problematic due to the often lengthy delay times caused by this separation. If the choir microphones are fed into a personal monitor system worn by the organist, the time delay is eliminated, and the organist is able to keep in sync with the choir. For pastors who desire a monitor, in-ears are also a viable option. Lavalier microphones, as well as gooseneck microphone found on pulpits, are especially prone to feedback issues with a floor monitor due to their increased sensitivity and greater distance from the sound source. A personal monitor will eliminate those concerns. The advantages extend well beyond the benefits to the performer, and increase the overall quality of the service and the worship experience.



CHAPTER SIX

EXPANDING THE PERSONAL MONITOR SYSTEM

Personal Monitor Mixers

Personal monitoring gives the performer an unprecedented level of control. But for the performer who desires more than simple volume and pan operation, an additional mixer can be implemented. Personal monitor mixers are especially useful for bands who have a limited number of available monitor mixes, or who do not have a monitor engineer, or anyone at all to run sound. In a perfect world, all performers would be happy listening to the exact same mix; in reality, everyone may want to hear something different. A small mixer located near the performers allows them to customize their mix to hear exactly what they desire. Theoretically, any mixer can double as a personal monitor mixer, but most lack one key feature; the input signals need to find their way to the main (FOH) mixer somehow. Large sound systems with separate monitor consoles use transformer isolated splitters to send the signals to two places, but these are prohibitively expensive for most working bands and small clubs. Y-cables can be used to split microphone signals, but they can get messy and are somewhat unreliable. A few manufacturers have recently introduced mixers with integrated microphone splitters (see figure 15). These can range from basic four channel mixers with volume and pan controls to larger, more fully-featured monitor consoles.

front panel

back panel (with split outputs)

Distributed Mixing

Distributed mixing is the direct result of advances in the area of digital audio networking. By converting analog audio signals to digital, audio can be routed to many locations without degradation or appreciable signal loss. Unlike with analog personal mixers, cabling is far simpler. Typically, analog outputs from a mixing console connect to an analog-to-digital converter. Multiple channels of digital audio can then be routed from the A-to-D converter to personal mixing "stations" located by each performer, using a single common Ethernet (Cat-5) cable, thus eliminating a "rat's nest" of microphone cables or the large, unwieldy cable snakes required for analog audio distribution. Cat-5 cable is inexpensive and readily available. The mixing station provides an analog headphone output that can drive a set of isolating earphones directly, or better yet, connect to either a hardwired or wireless personal monitor system. If nothing else, the personal monitor system offers the advantage of a limiter for some degree of hearing protection, as well as a volume control at the performer's hip. The mixers supplied with most distributed systems do not typically have a limiter. Most systems provide 8 to 16 channels of audio, allowing each performer to create their own custom mix, independent of other performers and without the intervention of a sound engineer. Note that giving this level of control to the performers will probably require some training in the basics of mixing to be successful (see sidebar "Creating a basic monitor mix" in Chapter 5).

Supplementary Equipment

In-ear monitoring is a different auditory experience from traditional stage monitoring. Since your ears are isolated from any ambient sound, the perception of the performance environment changes. There are several other types of audio products that can be added to a personal monitor system to enhance the experience, or try to simulate a more "live" feel.

Drum Throne Shakers

Something performers may miss when making the transition to personal monitors are the physical vibrations created by amplified low frequency sounds. Drummers and bass players are particularly sensitive to this effect. Although using a dual driver earphone usually results in more perceived bass, an earphone cannot replicate the physical sensation of air moving (sound) anywhere but in the ear canal. Drum shakers exist not to provide any audible sound reinforcement, but to recreate the vibrations normally produced by subwoofers

Figure 16: Drum Throne Shaker

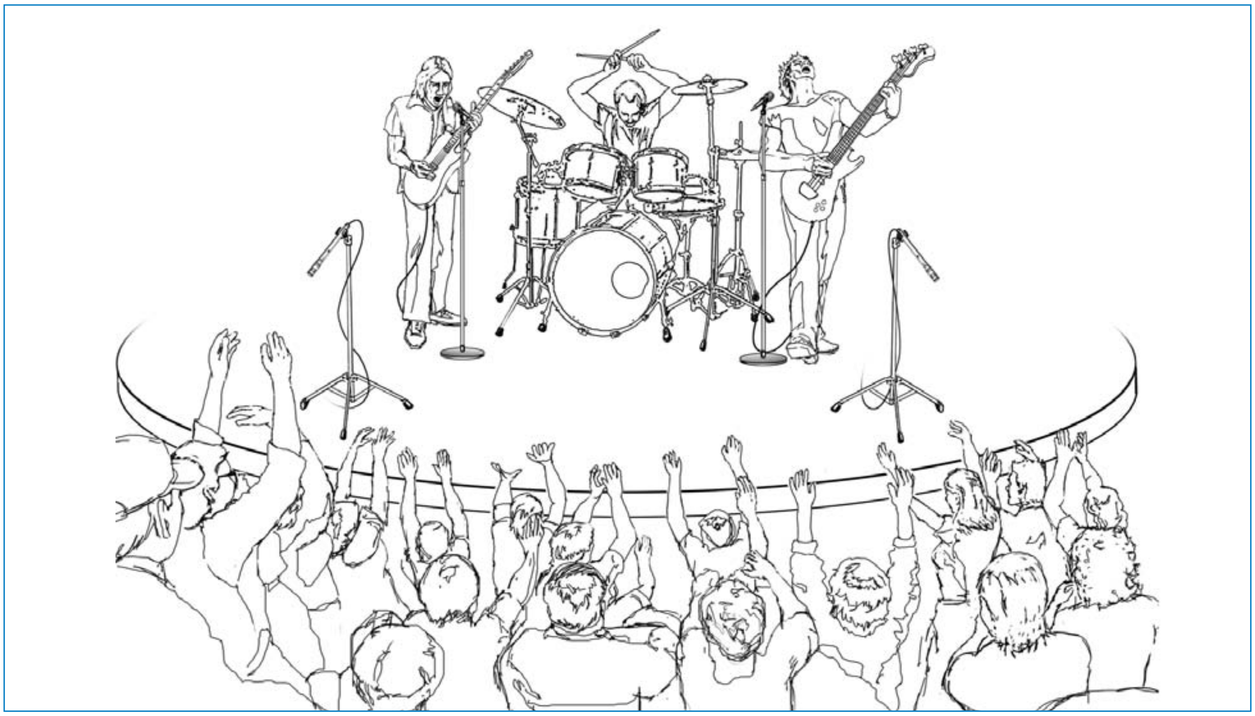


Figure 17: Ambient Microphones

or other low frequency transducers. Commonly found in car audio and cinema applications, these devices mechanically vibrate in sympathy with the musical program material, simulating the air disturbances caused by a loud subwoofer. They can be attached to drum thrones or mounted under stage risers (see figure 16).

Ambient Microphones

Ambient microphones are occasionally employed to restore some of the "live" feel that may be lost when using personal monitors. They can be used is several ways. For performers wishing to replicate the sound of the band on stage, a couple of strategically placed condenser microphones can be fed into the monitor mix. Ambient microphones on stage can also be used for performers to communicate with one another, without being heard by the audience. An extreme example (for those whose budget is not a concern) is providing each performer with a wireless lavalier microphone, and feeding the combined signals from these microphones into all the monitor mixes, but not the main PA. Shotgun microphones aimed away from the stage also provide good audience pick-up, but once again, a good condenser could suffice if shotguns are not available (see figure 17). The best solution is to use ambient earphones (see Chapter 4).

Effects Processing

Reverberant environments can be artificially created with effects processors. Even an inexpensive reverb can add depth to the mix, which can increase the comfort level for the performer. Many singers feel they sound better with effects on their voices, and personal monitors allow you to add effects without disturbing the house mix or other performers.

Outboard compressors and limiters can also be used to process the audio. Although many personal monitor systems have a built-in limiter, external limiters will provide additional protection from loud transients. Compression can be used to control the levels of signals with wide dynamic range, such as vocals and acoustic guitar, to keep them from disappearing in the mix. More advanced monitor engineers can take advantage of multi-band compression and limiting, which allows dynamics processing to act only on specific frequency bands, rather than the entire audio signal.

"In-ear monitor processors" combine several of these functions into one piece of hardware. A typical "in-ear processor" features multi-band compression and limiting, parametric equalization, and reverb. Secondary features, such as stereo spatialization algorithms that allow for manipulation of the stereo image, vary from unit to unit.

Latency and Personal Monitoring

An increasing number of devices used to enhance the personal monitor system are digital instead of analog. While the advantages of digital are numerous, including more flexibility and lower noise, any digital audio device adds a measurable degree of latency to the signal path, which should be of interest to personal monitor users. Latency, in digital equipment, is the amount of time is takes for a signal to arrive at the output after entering the input of a digital device. In analog equipment, where audio signals travel at the speed of light, latency is not a factor. In digital equipment, however, the incoming analog audio signal needs to be converted to a digital signal. The signal is then processed, and converted back to analog. For a single device, the entire process is typically not more than a few milliseconds.

Any number of devices in the signal path might be digital, including mixers and signal processors. Additionally, the signal routing system itself may be digital. Personal mixing systems that distribute audio signals to personal mixing stations for each performer using Cat-5 cable (the same cable used for Ethernet computer networking) actually carry digital audio. The audio is digitized by a central unit and converted back to analog at the personal mixer. Digital audio snakes that work in a similar manner are also gaining popularity.

Since the latency caused by digital audio devices is so short, the signal will not be perceived as an audible delay (or echo). Generally, latency needs to be more than 35 ms to cause a noticeable echo. The brain will integrate two signals that arrive less than 35 ms apart. This is known as the Haas Effect, named after Helmut Haas who first described the effect. Latency is cumulative, however, and several digital devices in the same signal path could produce enough total latency to cause the user to perceive echo.

Isolating earphones are the preferred type for personal monitors, because they provide maximum isolation from loud stage volume. Isolating earphones, however, result in an effect known as the occluded ear (see Chapter 4). Sound travels by at least two paths to the listener's ear. The first is a direct path to the ear canal via bone conduction. An isolating earphone reinforces this path, creating a build-up of low frequency information that sounds similar to talking while wearing earplugs. Secondly, the "miked" signal travels through the mixer, personal monitor transmitter and receiver, and whatever other processing may be in the signal path. If this path is entirely analog, the signal travels at the speed of light, arriving at virtually the same time as the direct ("bone-conducted") sound. Even a small amount of latency introduced by digital devices, though, causes comb filtering.

Before continuing, an explanation of comb filtering is in order. Sound waves can travel via multiple paths to a common receiver (in this case your ear is the receiver). Some of the waves will take a longer path than others to reach the same point. When they are combined at the receiver, these waves may be out of phase. The resultant frequency response of the combined waves, when placed on a graph, resembles a comb, hence the term comb filtering.

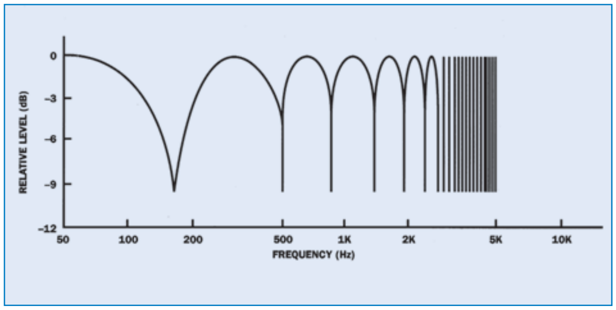


Figure 18: Comb Filtering

"Hollow" is a word often used to describe the sound of comb filtering.

It is generally believed that the shorter the latency, the better. Ultimately, changing the amount of latency shifts the frequency where comb filtering occurs. Even latency as short as 1 ms produces comb filtering at some frequencies. What changes is the frequency where the comb filtering occurs. Lower latency creates comb filtering at higher frequencies. For most live applications, up to 2 ms of delay is acceptable. When using personal monitors, though, total latency should be no more than .5 ms to achieve sound quality equivalent to an analog, or zero latency, signal path. While in reality it may be difficult to achieve latency this short, be aware that any digital device will cause some latency. The individual user will have to determine what amount of latency is tolerable. As an alternative, some users report that inverting the polarity of certain input channels, or even the entire mix, improves the sound quality. Keep in mind that comb filtering still occurs, but at frequencies that may be less offensive to the listener.

CHAPTER SEVEN

WIRELESS BASICS

As mentioned previously, wireless equipment by nature adds an extra level of complexity over a hardwired system. When considering wireless personal monitors (or wireless microphones), a basic understanding of wireless concepts (and the behavior of radio waves) will help ensure consistent, reliable performance.

Radio Waves

Most wireless audio equipment accomplishes its task by converting the audio signal into a radio wave, then back to an audio signal. The technical definition of a radio wave is a series of electromagnetic field variations traveling through space (see figure 19). Radio waves travel at the speed of light and are able to travel a significant distance from the source.

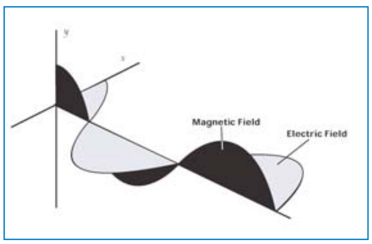


Figure 19: Radio Wave

These characteristics make radio the ideal form of transmission for audio applications, from broadcast radio and television to cordless phones. Radio waves also do not require a medium for travel (sound waves need air to propagate); they can even travel through the vacuum of space.

Like sound, a radio wave can be described by its frequency and its amplitude. Frequency is measured in Hertz (cycles per second). Frequencies in the radio spectrum range from a few Hertz to beyond the Gigahertz (GHz) range (see figure 20). Most professional wireless audio systems operate in the Megahertz (MHz) range. Humans are only directly sensitive to radio waves in the range of a few million Gigahertz, which we perceive as visible light, and those frequencies just below visible light, which are perceived as heat (infrared radiation). Radio waves such as those produced by wireless microphones surround us on a daily basis (think of all the radio stations on the air!), yet we are oblivious to their existence until you turn on the radio. Amplitude refers to the "strength" of the radio wave, and is typically measure in volts per meter. Finally, wavelength is the physical distance between the start of one wave cycle and start of the next cycle as the wave moves through space. Wavelength is directly related to frequency in that lower frequencies have longer wavelengths and higher frequencies have shorter wavelengths (see figure 21). Wavelength is important in antenna design as well as in how the wave will behave when presented with an object in its path.

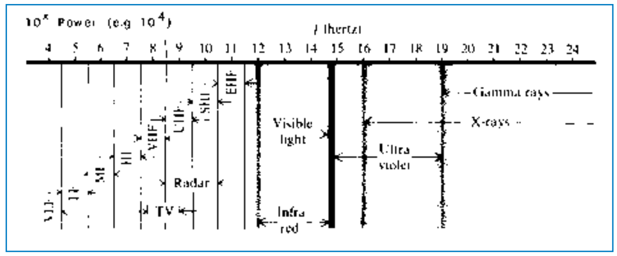


Figure 20: Radio Frequency Spectrum

Metal objects physically larger than the radio wave reflect the signal. Practically, metal objects should never be placed between the transmitter and receiver, or reception will suffer. Interestingly, a reflecting metal object can be porous, i.e. it can have holes or spaces in it. If the holes are much smaller than the wavelength, the metal surface will behave as if it were solid. A screen door, for example, would reflect an RF signal created by most wireless microphone transmitters, which typically have fairly long wavelengths. Non-metallic substances (including air) do not reflect waves, but are not completely transparent either. They generally "attenuate" or cause a loss in the strength of radiowaves that pass through them. While a brick wall will not reflect a radio wave, the signal passing through it will most likely be at a much lower level. This argues strongly for line of sight between transmitter and receiver, line of sight meaning that if the transmitter antenna had eyes, it could see the receiver antenna. Human bodies, which consist mostly of water, are excellent attenuators. Therefore, people as well as walls can obstruct line of sight.

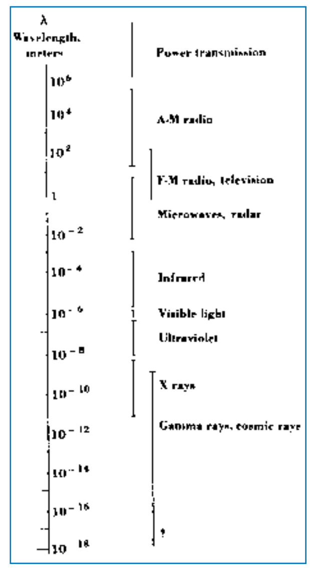


Figure 21: Radio Wavelength Chart

Radio "information" is generally transmitted using only one frequency. The wave itself can be varied in several different ways that allow it to "carry" the desired information; in this case the monitor mix. The most common ways to vary the radio wave are AM and FM. AM, which stands for Amplitude Modulation, varies the strength, or amplitude of the signal (see figure 22). If the frequency is varied, then it is called Frequency Modulation or FM (see figure 23). All wireless personal monitor systems and wireless microphones

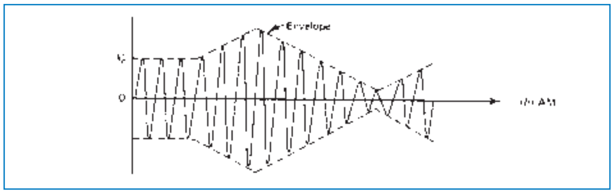


Figure 22: Modulation AM Carrier

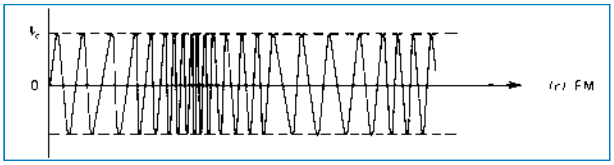


Figure 23: Modulation FM Carrier

use the FM technique. More "information" can be sent in a typical FM signal, which allows for higher fidelity and less sensitivity to common sources of radio noise (such as lightning and electrical power equipment). Listen to an AM radio station versus an FM station for a convincing comparison.

Antennas

The function of an antenna is to act as the interface between the internal circuitry of the transmitter (or receiver) and the external radio signal. Antennas for wireless audio applications typically fall into three categories, 1/4 wave, 1/2 wave, and unidirectional. The minimum size for adequate reception or transmission is a 1/4 wave antenna, named because its length is approximately equal to 1/4 of the wavelength of the desired signal. The most important consideration in the performance of a 1/4 wave antenna is the presence of a "ground plane", a metal surface at least 1/4 wavelength long and electrically connected to the transmitter or receiver ground. In most applications the transmitter or receiver itself acts as a ground plane. 1/4 wave antennas, therefore, cannot be remote mounted. They must be attached to a ground plane for efficient operation. A properly designed 1/2 wave antenna does not require a ground plane, allowing it to be remotely mounted with relative ease. 1/2 wave antennas also provide a theoretical increase in sensitivity of up to 3 dB greater than the 1/4 wave antenna in some configurations.

Unidirectional antennas are also available for wireless. They are often referred to as "paddle" or "shark fin" antennas, since they usually consist of a flat surface that resembles a ping-pong paddle, with a horizontal boom and multiple transverse elements mounted directly to the surface. They can achieve high gain (up to 10 dB compared to the 1/4 wave type) in one direction and reduce multipath drop outs (drop outs that occur when radio waves bounce around the room and arrive at the receiving antenna out of phase). Unidirectional antennas are typically employed to increase the operating range of the system. Additionally, they also do not require a ground plane for effective operation. For maximum efficiency, receiving antennas should be oriented in the same direction as the transmitting antenna. For example, the receiving antenna should be vertical if the transmitting antenna is vertical.

Range of Wireless Systems

A logical question concerning wireless performance is the effective operating range of various systems. Unfortunately, the answer is much more complicated than a simple distance measurement. The receiver must be able to pick up a "useable" signal from the transmitter. "Useable" means the signal is strong enough for the receiver to recognize, as well as sufficiently stronger than any background RF noise or other potential interfering signals. In most urban locations, even if the frequency of the wireless system in question is clear of any direct interference, the overall level of general RF noise may, at some point, be the limiting factor is how much range is available for a given system. Other factors important to operating range include transmitter output power (usually stated in milliwatts – mW), antenna efficiency, receiver sensitivity, and the ability of the receiver to reject unwanted signals and noise.

Rather than quote a specific maximum operating distance, most manufacturers of wireless systems give a "typical" range. For systems of the type discussed here (50 or 100 mW, VHF or UHF) the typical range may vary from 100 feet to 1000 feet. The lower number represents a moderately severe environment while the upper figure might only be achieved in absolute ideal conditions. Extremely poor conditions could result in a range of only 50 feet or less. It is impossible to accurately predict the range of an arbitrary system in an arbitrary application.

Stereo Wireless Transmission

Many microphones and most circuitry used in the reproduction of audio signals have a bandwidth of 20 kHz (20 Hz - 20 kHz) or more. Digital devices operating at a sampling rate of 44.1 kHz have frequency responses extending to 22 kHz; 48 kHz sample rates are flat out to 24 kHz. Many boutique analog devices boast flat response beyond 30 and sometimes 40 kHz! Undoubtedly, these are great advances in the sound reproduction field, unless audio with that kind of bandwidth is sent through the air in the form of a stereo encoded wireless transmission. It's the equivalent of ten pounds of audio in a five-pound bag.

Every so often, a performer will complain of an ear mix that just doesn't sound "quite right," no matter what adjustments are made to levels or EQ. Sometimes it's a simple image shift; sometimes it is distortion that happens at the strangest times. The output of the mixing console sounds fine, as does the headphone output of the transmitter. Changing frequencies, cables, earpieces, and bodypacks makes no difference. Ultimately, respecting the frequency response limitations of stereo wireless transmission is the key to successfully creating stable, good-sounding ear mixes. There are several ways this can be accomplished, but as a general rule, avoid any frequency boosts above 15 kHz. Stereo multiplexed wireless transmission has a limited frequency response of 50Hz - 15 kHz. This frequency response limitation has been in place since the FCC approved stereo multiplexed transmissions (MPX) back in 1961. Audio engineers mixing stereo wireless transmissions for on-stage talent wearing personal monitors should be aware of the operating principals of MPX stereo to achieve the desired results at the receiver.

In many cases, switching to a mono transmission clears up any wireless anomaly (except for interference) in these types of monitoring systems. However, many users want to monitor in stereo, so being aware of the limitations of MPX encoding will allow for greater talent satisfaction.

Stereo wireless transmitters use a steep cut filter, or “brick-wall filter,” prior to modulation, centered at 19 kHz to create a safe haven for the required “pilot tone.” MPX encoders in stereo wireless transmitters use a 19 kHz pilot tone to inform receivers that the transmission is encoded in stereo. If the receiver does not sense a 19 kHz pilot tone, it will only demodulate a mono signal. Moreover, if the 19 kHz pilot tone is not stable, stereo imaging degrades at the receiver. Most importantly, if personal monitor receivers do not sense stable 19 kHz pilot tones, they will mute (this is called tone-key squelch, a circuit designed to keep the receiver muted when the corresponding transmitter is turned off.). Problems are created due to the extensive EQ capabilities of modern mixing consoles, which offer high frequency shelving equalization from as low as 10 kHz to as high 12, 15 and even 16 kHz. Digital mixing consoles offer parametric filtering that can center on practically any frequency and boost by as much as 18 dB. With a multi-channel mixing board, it is easy enough to create a counteractive frequency response at the frequency of interest - 19 kHz. In stereo wireless, there are two pieces of information actually being transmitted, the “mono” or “sum” signal (Left + Right) and the “difference” (Left - Right) channel, each occupying a 15 kHz-wide swath of spectrum. The 19 kHz pilot tone is centered in between these two signals (see figure 24).

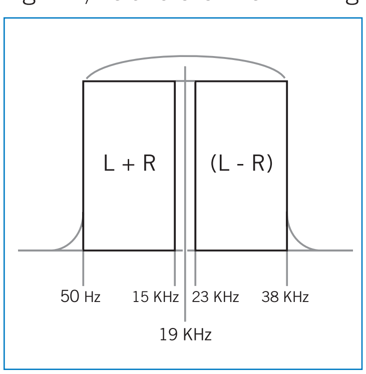


Figure 24: Stereo Pilot Tone

The stereo image is restored in the receiver by adding the sum and difference signals to create the left channel, and subtracting them to derive the right channel.

$$(L + R) + (L - R) = 2L$$
$$(L + R) - (L - R) = 2R$$

This system ensures mono compatibility, since the received signal will simply collapse to mono when the pilot tone is lost. You are left with nothing but the L + R “sum” signal.

However, since the 19 kHz pilot tone resides in the audio band, it can easily be compromised by the program material. The result of these high frequency components getting into the modulator can cause, at best, degradation of stereo separation and distortion, and in worst-case situations, muting of the receiver. Add the high frequency shelf used in the pre-emphasis curves prior to the companding circuits in stereo transmitters (a form of noise reduction), and it is easy to see how a small high frequency boost on a channel strip can have a huge effect on what is heard after the RF link. These phenomena are usually blamed on the product in question and easily dismissed as “bad frequency selection” (interference), “multi-path drop out” or “dying batteries”, when in reality these anomalies are the sound of stereo multiplex transmission breaking down at its fundamental operating principal. If audio modulates the pilot tone, stereo reception and the resultant sound quality will be poor. If upper harmonics of musical instruments aggravate the (L-R) sidebands (especially in a transient manner - tambourines, triangles, high hats, click tracks, etc.), stereo separation can degrade, frequency response can be compromised and even dynamic interactions between one channel and another can be detected.

Sensible use of the pan knobs on stereo sources can turn a “difficult to receive” (and encode!) signal into a much more “easily received” (and decoded!) signal. Instead of panning hard left and right, try the “10 o'clock” and “2 o'clock” positions. Judicious equalization prior to stereo transmission can pre-shape a mix for smoother MPX encoding. *Ultimately, to avoid disturbing the pilot tone, use 1/10th of an octave notch filters at 16 kHz on console output busses to increase the slope of the MPX filter.* Moreover, a nominal balance between transient and sustained signals in both the frequency response and stereo perspective domains of a mix are useful tools to create realistic environments inside performers' heads.

Antenna Types

As of this writing, professional personal monitor receivers are of the single-antenna, non-diversity type, making them subject to a phenomenon known as multi-path dropout. Most personal monitor systems are supplied with an omnidirectional transmit antenna, which radiates the signal in all directions. Dropouts occur as radio waves reflect around the performance area, traveling multiple paths to the receiver antenna. In certain locations, these waves will arrive out of phase, canceling each other out, resulting in an audible drop out. Wireless microphones overcome this problem by employing diversity technology in the receivers, but this feature is not yet readily available in most personal monitor receivers. Unidirectional antennas provide one possible way to minimize multipath dropouts. When used as a transmit antenna, the radiation pattern of the unidirectional types concentrate the RF signal in one direction, minimizing the reflections that can cause multipath. Unidirectional antennas typically provide up to 7 dB of additional forward gain, which improves performance when the distance from transmitter to receiver is greater than 50 feet. Log periodic antennas are the most common type of unidirectional antenna used in professional audio applications.

Both omni- and unidirectional antennas are polarized, that is, the horizontal or vertical orientation of the antenna is crucial to obtaining optimum pickup of the radio wave. Ideally, the receiving antenna should be oriented in the same plane as the transmitting antenna. However, in a live performance, the receiver antenna often ends up “out of polarity” with the transmit antenna as the performer moves around. A third type of antenna, helical (see figure 25), is not sensitive to polarity. Therefore, no matter the position of one antenna, the helical antenna (whether transmitting or receiving) will always achieve maximum signal transfer, further reducing the chance of dropouts due to multi-path interference. Beyond the polarity issue, helical antennas offer greater directionality than log periodic types and approximately 14 dB of forward gain versus an omnidirectional antenna. See the chart below for a comparison of the various antenna types.

Figure 25: Helical Antenna
(Image Courtesy of PWS)

Antenna Type	*Directionality*	*Forward Gain*	*Polarity Sensitive?*
Omni	360 degrees	0 dB	Yes
Log periodic	120 degrees	7-8 dB	Yes
Helical	57 degrees	14 dB	No

Antenna Combining

Antenna combining is crucial to obtaining optimal RF performance from personal monitor transmitters. Several closely-spaced, high-power transmitters can produce excessive intermodulation products (a transmitter interaction that produces additional frequencies), which in turn can cause additional dropouts at the receiver. In this case, a passive combiner should be used for combining two transmitters. For more than two, though, an active combiner is recommended. An active antenna combiner typically accepts between 4 to 8 transmitters. Unlike active antenna distribution systems used for wireless microphone receivers, which can be cascaded together for larger setups, active antenna combiners should never be “actively” cascaded. If more than one combiner is needed to combine all the transmitters together, a passive combiner should be used to connect two active combiners together. As always, be aware of any extra signal loss incurred with the passive combiners.

Similar to active antenna distribution systems, active combiners also have a specified frequency bandwidth. Be sure to select the proper bandwidth for the given transmitter frequencies.

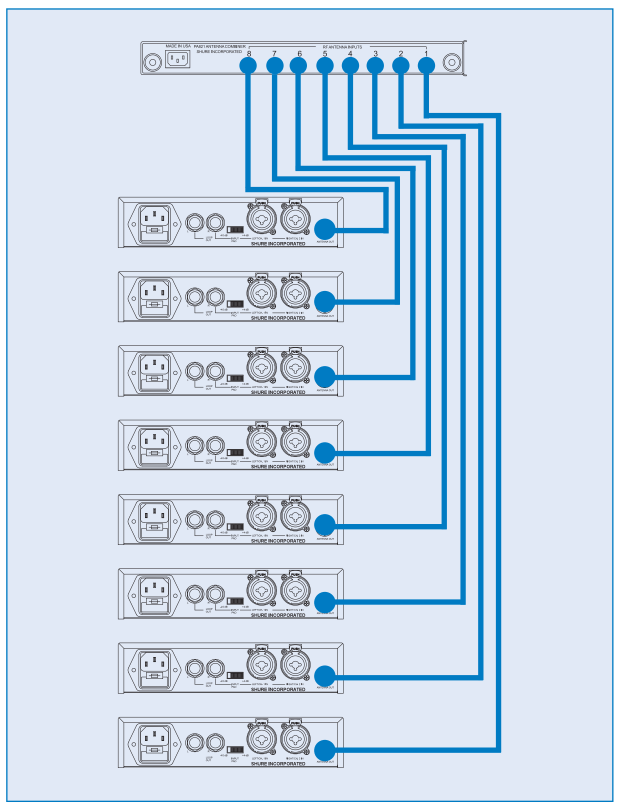
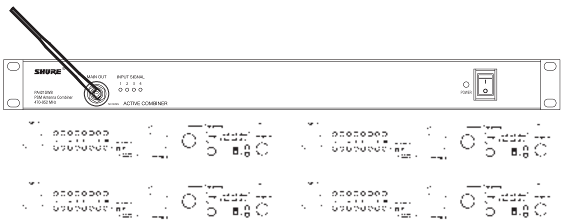


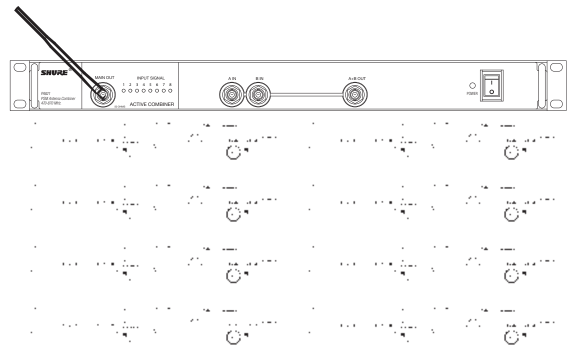
Figure 26: Active Antenna Combining



Antenna combining: 2-4 systems

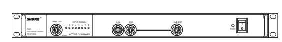
(1) 4-to-1 antenna combiner

Shown: Shure PA421 antenna combiner

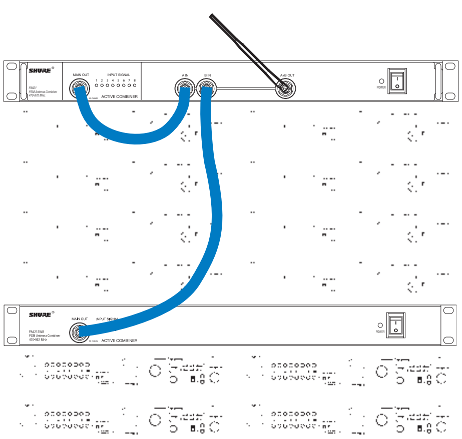


Antenna combining: 5-8 systems

(1) 8-to-1 active combiner

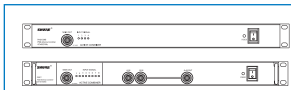


Shown: Shure PA821 with built-in 2-to-1 passive combiner



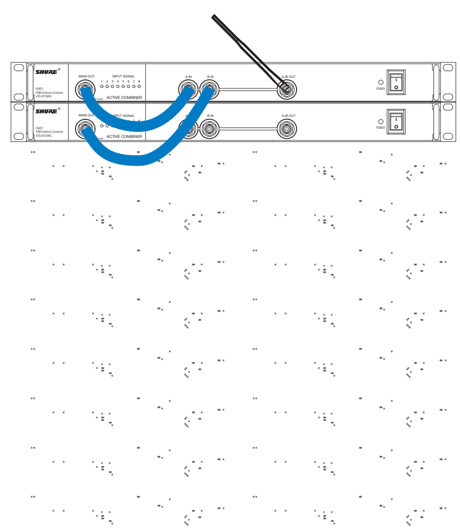
Antenna combining: 9-12 systems

- *(1) 8-to-1 active combiner with 2-to-1 passive combiner*
- *(1) 4-to-1 active combiner*



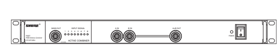
Shown: Shure PA421 antenna combiner and PA821 antenna combiner with built-in 2-to-1 passive combiner

** For 9 systems, 4 input combiner not needed.*



Antenna combining: 13-16 systems

(2) 8-to-1 active combiners with 2-to-1 passive combiner



Shown: Shure PA821 with built-in 2-to-1 passive combiner

Frequency Coordination

A complete discussion of frequency coordination is beyond the scope of this publication, but it is important to note that just like wireless microphones, frequencies for wireless personal monitors systems should not be chosen at random. When multiple transmitters are used in the same location, they interact with one another to produce additional frequencies know as intermodulation products (IMD). Frequencies must be chosen such that they avoid IMDs, or interference will result. Wireless microphone systems, wireless intercoms, and any other devices that transmit in the UHF frequency band must be considered when coordinating frequencies. Manufacturers should be able to assist with frequency coordination. Additionally, software packages that can quickly and accurately calculate frequency compatibility are highly recommended.

When choosing a wireless personal monitor system, remember that more mixes require more transmitters, and the system must be able to handle that number of transmitters on the air simultaneously. More on-air transmitters usually require a higher priced system; inexpensive systems tend to be limited in the number of compatible channels available.

Avoiding local television broadcasts is another important ingredient in successful operation. Again, just like wireless microphones, wireless monitors operate in the same spectrum used by over-the-air television. The power levels produced by commercial television broadcast transmitters are many orders of magnitude higher than those of professional audio devices, and will significantly disrupt their operation. As a general rule, avoid any occupied television channels within 50 miles of the venue where wireless monitors will be used. Most wireless personal monitor systems feature some degree of frequency agility, which allows the user to change the operating frequency to avoid interference. Additionally, keep in mind that television channels are different in every city. For touring applications, be certain to choose a frequency agile system so a new frequency can be selected when necessary.

Coordinating frequencies?
Looking for unoccupied television channels?

Check out:
www.shure.com/frequency or www.shure.com/faq

When choosing a wireless system, be it a microphone or personal monitors, there are three questions that need to be asked:

1. Where will the wireless system be used?

2. How many wireless systems will be used?

3. Are there any other existing wireless systems (microphones included) currently in use?

Tips for Effective Operations

1. **Maintain line-of-sight between the transmitter and receiver antennas as much as possible.**
   Avoid metal objects, walls, and large numbers of people between the receiver and its associated transmitter. Ideally, this means that transmitter antennas should be located in the same room as the receivers, and elevated above the audience or other obstructions.

2. **Keep the transmitter antenna a reasonable distance to the receiver.**
   Maximum distance is limited by transmitter power, interference, and objects that block line-of-sight. Keeping the distance between transmitter and receiver antennas short will minimize the effects of these other factors, but beware of getting too close. A minimum distance of 10 feet is recommended to avoid detrimental interference caused by interactions in the receiver (called intermodulation).

3. **Use the proper type of antenna.**
   A $^1/_4$-wave antenna can be used if it is attached directly to the receiver, to an antenna combiner, or to another panel which acts as a ground plane. If the antenna is to be located at a distance from the receiver a $^1/_2$-wave antenna is recommended. For applications requiring more distant antenna placement, it may be necessary to use a directional antenna.

4. **Locate transmitters away from any suspected sources of interference.**
These include digital equipment (CD players, digital effects processors, computers), AC power equipment, lighting dimmers and controllers. It is generally considered good practice to keep wireless personal monitor transmitters separate from wireless microphone receivers. A relatively high power transmitter located next to a fairly sensitive microphone receiver can desensitize the receiver, causing poor performance from the wireless microphone. A distance of a few feet is typically adequate.

5. **Use an antenna combiner when possible.**
When using multiple transmitters (3 or more), an antenna combiner can help improve performance. Combining all transmitters down to one antenna greatly reduces dropouts due to multipath interference, and also eliminates unwieldy antenna "farms." Active antenna combiners should never be cascaded one into another, as this aggravates intermodulation problems. A passive combiner should be used when multiple active combiners are required to accommodate a large number of transmitters.

6. **Verify good batteries in all receivers.**
Always use fresh batteries of the correct type in the receiver. Most manufactures recommended only alkaline type batteries for proper operation. Alkalines operate up to 10 times longer than so-called "heavy duty" non-alkaline types. Buy in bulk quantities to save money. Use rechargeable batteries with extreme caution: their power capacity is much lower than the same size alkaline and their initial voltage is usually less. Typical "9 volt size" Ni-Cad rechargeables have an initial voltage of only 7.2 volts. Practically, this translates to an operating time as low as 15 minutes in some units. Experienced users always choose alkaline batteries.

7. **Select clear operating frequencies.**
This process varies slightly depending on the unit. Start with all receivers on, and corresponding transmitters off. If the RF indicators are illuminated on any receiver, this indicates the presence of an interfering signal; change the frequency if possible. Some units have built in frequency selection modes that can aid in this process. Secondly, make sure all frequencies are compatible with each other. This will vary among manufacturers, check the supplied documentation to assist in determining frequency compatibility.

8. **Perform a listening test**.
Walk around the performance area with each individual system while all systems are on to verify no audible interference or dropouts.

CHAPTER EIGHT

SAFE LISTENING WITH PERSONAL MONITORS

No discussion of monitoring systems would be complete without some discussion of human hearing. The brain's ability to interpret the vibration of air molecules as sound is not entirely understood, but we do know quite a bit about how the ear converts sound waves into neural impulses that are understood by the brain.

The ear is divided into three sections; the outer, middle, and inner ear (see figure 27). The outer ear serves two functions – to collect sound and act as initial frequency response shaping. The outer ear also contains the only visible portion of the hearing system, the pinna. The pinna is crucial to localizing sound. The ear canal is the other component of the outer ear, and provides additional frequency response alteration. The resonance of the ear canal occurs at approximately 3 kHz, which, coincidentally, is right where most consonant sounds exist. This resonance increases our ability to recognize speech and communicate more effectively. The middle ear consists of the ear drum and the middle ear bones (ossicles). This section acts as an impedance-matching amplifier for our hearing system, coupling the relatively low impedance of air to the high impedance of the inner ear fluids. The ear drum works in a similar manner to the diaphragm of a microphone, it moves in sympathy to incoming sound waves, and transfers those vibration to the ossicles. The last of these bones, the stapes (or stirrup, as our grade school teachers called it), strikes an oval-shaped window that leads to the cochlea, the start of the inner ear. The cochlea contains 15,000 to 25,000 tiny hairs, known as cilia, which bend as vibrations disturb the fluids of the inner ear. This bending of the cilia sends neural impulses to the brain via the auditory nerve, which the brain interprets as sound. Hearing loss occurs as the cilia die. Cilia begin to die from the moment we are born, and they do not regenerate. The cilia that are most sensitive to high frequencies are also the most susceptible to premature damage. Three significant threats to cilia are infection, drugs, and noise. Hearing damage can occur at levels as low as 90 db SPL. (see figure 28.) According to OSHA (Occupational Safety Health Administration), exposure to levels of 90 dB SPL for a period of 8 hours could result in some damage. Of course, higher levels reduce the amount of time before damage occurs. Hearing conservation is important to everyone in the audio industry. As mentioned before, a personal monitor system can assist in helping to prevent hearing damage – but they are not foolproof protection. The responsibility for safe hearing is now in the hands of the performer. At this point, there is no direct correlation between where to set your volume control and how much SPL is present in your ears. Here are a few suggestions, though, to help protect your hearing.

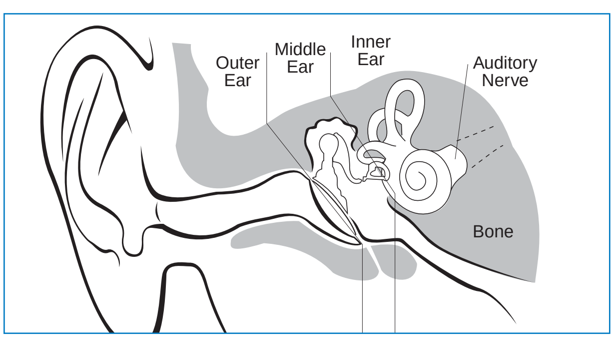


Figure 27: Ear Anatomy

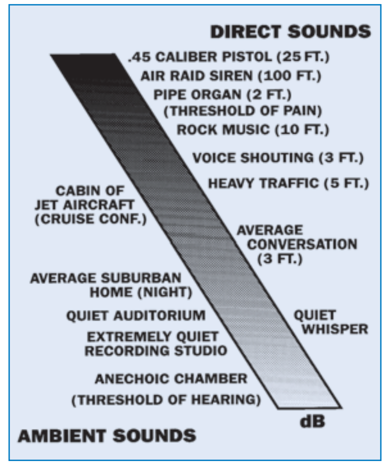


Figure 28: Sound Pressure Level of Typical Sources

Use an isolating earphone

Without question, the best way to protect yourself from high sound pressure levels is to use a high quality earplug. The same reasoning applies to a personal monitor earphone. When using personal monitors, listening at lower levels requires excellent isolation from ambient sound, similar to what is provided by an earplug. Hearing perception is based largely on signal-to-noise. To be useful, desired sounds must be at least 6 dB louder than any background noise. Average band practice levels typically run 110 dB SPL, where hearing damage can occur in as little as 30 minutes. To hear your monitors when using a non-isolating earphone would require a sound level of 116 dB SPL, which reduces the exposure time to 15 minutes! Inexpensive "ear buds", like those typically found with "Walkman"-type personal radios, offer little, if any, isolation. Avoid these types of earphones for personal monitor applications. Not all types of isolating earphones truly "isolate", either. Earphones based on dynamic drivers typically require a "ported" enclosure to provide adequate low frequency response. This "port", a small hole or multiple holes in the enclosure, drastically reduces the effectiveness of the isolation. Note that not all dynamic earphones require ports. Some designs use a sealed, resonating chamber to accomplish the proper frequency response, thus negating the need for ports but preserving the true isolating qualities of the earphone. Earphones that employ a balanced armature transducer, similar to those used in hearing aids, are physically smaller and do not require ports or resonating chambers. In fact, balanced armature-type earphones rely on a good seal with the ear canal to obtain proper frequency response. They can be made somewhat smaller, but are typically more expensive, than their dynamic counterparts.

Use both earphones!

A distressing, yet increasingly common, trend is only using one earphone and leaving the other ear open. Performers have several excuses for leaving one ear open, the most common is a dislike for feeling "removed" from the audience, but the dangers far outweigh this minor complaint. First, consider the above example of a 110 dB SPL band practice. One ear is subjected to the full 110 dB, while the other ear needs 116 dB to be audible. Using only one earphone is equivalent to using a non-isolating earphone, except one ear will suffer damage twice as fast as the other! Second, a phenomenon known as binaural summation, that results from using both earphones, tricks the ear-brain mechanism into perceiving a higher SPL than each ear is actually subjected to. For example, 100 dB SPL at the left ear and 100 dB SPL in the right ear results in the perception of 106 dB SPL. Using only one earphone would require 106 dB SPL at that ear. The practical difference? Potential hearing damage in one hour instead of two hours. Using both earphones will usually result in overall lower listening levels.

The following table shows OSHA recommendations for exposure time versus sound pressure level:

Sound Pressure Level	Exposure time
90 dB SPL	8 hours
95 dB SPL	4 hours
100 dB SPL	2 hours
105 dB SPL	1 hour
110 dB SPL	30 minutes
115 dB SPL	15 minutes

Ambient microphones are commonly employed to help overcome the "closed-off" feeling. An ambient microphone can be a lavalier clipped to the performer and routed directly to the personal monitor mix, or a stereo microphone pointed at the audience. The common thread is that they allow the user to control the level of the ambience.

Keep the limiter on

Unexpected sounds, such as those caused by someone unplugging a phantom-powered microphone or a blast of RF noise, can cause a personal monitor system to produce instantaneous peaks in excess of 130 dB SPL, the equivalent of a gun shot at your ear drum. A brick-wall type limiter can effectively prevent these bursts from reaching damaging levels. Only use a personal monitor system that has a limiter at the receiver, and do not defeat it for any reason. A well-designed limiter should not adversely affect the audio quality, as it only works on these unexpected peaks. If the limiter seems to be activating too often, then the receiver volume is probably set too high (read as: unsafe!). Outboard compressors and limiters placed before the inputs of the monitor system are certainly appropriate, but are not a substitute for an onboard limiter, as they cannot protect against RF noise and other artifacts that may occur post-transmitter.

Pay attention to what your ears are telling you

Temporary Threshold Shift (TTS) is characterized by a "stuffiness," or compressed feeling, like someone stuck cotton in your ears. Ringing (or "tinnitus") is another symptom of TTS, which is the ear's way of telling you that you are being exposed to sound levels that are too extreme. Please note, though, that hearing damage may have occurred even if you do not experience ringing. In fact, the majority of people who have hearing damage never reported any ringing. After experiencing TTS, most of your hearing will recover. Permanent damage has possibly occurred, though. The effects of TTS are cumulative, so if you regularly experience the above effects, your monitoring level is too loud and hearing damage will occur with repeated exposure to those levels. Turn it down!

Have your hearing checked regularly

The only certain way to know if your listening habits are safe is to get your hearing checked regularly. The first hearing test establishes a baseline that all future hearing exams are compared against to determine if any loss has occurred. Most audiologists recommend that musicians have their hearing checked at least once a year. If hearing loss is caught early, corrections can be made to prevent further damage.

A frequently asked question about personal monitors is: "How do I know how loud it is?" At this time, the only way to develop a useful correlation between the volume knob setting and actual SPL at the ear drum is by measuring sound levels at the ear drum with specially made miniature microphones. A qualified audiologist (not all have the right equipment) can perform the measurements and offer recommendations for appropriate level settings.

Personal monitors can go a long way towards saving your hearing, but only when used properly. Monitoring at lower levels is the key to effective hearing conservation, and this can only be accomplished through adequate isolation. Used correctly, professional isolating earphones, combined with the consultation of an audiologist, offer the best possible solution for musicians interested in protecting their most valuable asset.

The beauty of personal monitors is the fact that you can turn it down. Not only do personal monitors offer improved sound quality and convenience, they put you back in control of your sound.

House Ear Institute	H.E.A.R.
Hotline: (213) 483-4431 Website: www.hei.org	Hotline: (415) 409-3277 Website: www.hearnet.com

Sensaphonics Hearing Conservation
660 N. Milwaukee Avenue, Chicago IL 60622
Toll Free: (877) 848-1714 In Chicago: (312) 432-1714
Fax: (312) 432-1738
Website: www.sensaphonics.com
E-mail: saveyourears@sensaphonics.com

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 the other voltage, and log is logarithm base 10.

Examples:

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

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 0.001 volt and 1 volt?

dB = 20 x log(0.001/1)
dB = 20 log(0.001)
dB = 20 x (-3) (the log of 0.001 is -3)
dB = -60

That is, 0.001 volt is 60dB less than 1 volt.

Similarly:

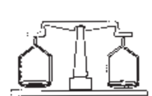
if one voltage is equal to the other they are 0dB different;

if one voltage is twice the other they are 6dB different;

if one voltage is ten times the other they are 20dB 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 decibel value such as: dBV (reference to 0.0002 microbar which is 0dB Sound Pressure Level).

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 1dB SPL is about the smallest difference in loudness that can be perceived while a 3dB SPL change is generally noticeable. A 6dB SPL change is quite noticeable and finally, a 10dB SPL change is perceived as "twice as loud."

1. Compare	2. Compress	3. Scale (x 20)
	10^0=1	0
	10^1=10	20
	10^2=100	40
	10^3=1000	60
b a	10^4=10,000	80
	10^5=100,000	100
b/a	10^6=1,000,000	120

Selection Guide

	PSM® 200	PSM® 400	PSM® 600	PSM® 700
Wireless Version	UHF 8 Selectable Frequencies across 6 TV Channels *Frequency Synthesized*	UHF 16 Selectable Frequencies *Frequency Synthesized*	UHF 2 Selectable Frequencies across 2 TV Channels *Crystal*	UHF 32 Selectable Frequencies Across 4 TV Channels *Frequency Synthesized*
Frequency Range	H2—518-554 MHz (TV 22-27)	X1-944-952 MHz	626-662 MHz (TV 40-45)	H3—524-554 (TV 23-27) L2—632-662 (TV 41-45)
Max Number of Transmitters in One Location[1]	4	6	10	20 (using both frequency bands)
Transmitter Output Power	30 mW	50 mW	100 mW	100 mW
Transmitter Input Level Control	Yes	No	Yes	Yes
Transmitter Input Connectors	Combination XLR female 1/4" phone jacks	Balanced 1/4" phone jacks	Combination XLR female 1/4" phone jacks	Combination XLR female 1/4" phone jacks
Transmitter Headphone Monitor Jacks	None	3.5 mm mini-phone jack	1/4" and 3.5 mm phone jacks	1/4" and 3.5 mm phone jacks
Transmitter Input Pad	No	No	Yes (-10 dBV or +4 dBu)	Yes (-10 dBV or +4 dBu)
Loop Through Outputs	Split Outputs	Yes	Yes	Yes
Receiver Chassis	High-Impact Plastic	High-Impact Plastic	Metal	Metal
Receiver Battery Type/Life[2]	9-volt alkaline 4 hours	9-volt alkaline 8 hours	9-volt alkaline 8 hours	9-volt alkaline 6 hours
Transmitter Antennas	Fixed 1/4-wavelength	Removable 1/4-wavelength	Removable 1/4-wavelength	Removable 1/4-wavelength
Rack Mount Hardware	Included	Included	Included	Included
Antenna Combiner	N/A (Antenna not removable)	PA421SWB	PA421/PA821	PA421/PA821
Antenna Cable	N/A	PA725 (10 ft.)	PA725 (10 ft.)	PA725 (10 ft.)
Remote Directional Antenna	N/A	PA805SWB/PA805X	PA705	PA805SWB
Transmitter Power Supply	PS20	PS40	Internal	Internal
Hardwired Version	Yes	Yes	Yes	No
Hardwired Chassis Type	High-Impact Plastic	High-Impact Plastic	Metal	N/A
Hardwired Battery Type/Life[2]	9-volt alkaline 6 hours	9-volt alkaline 8 hours	9-volt alkaline 10 hours	N/A

[1]Location dependent [2]Battery life is volume level dependent

Ambience – Room acoustics or natural reverberation.

Amplitude – Magnitude of strength of signal or wave.

Antenna – Electrical circuit element that transmits or receives radio waves.

Auxiliary (Aux) Send – An extra output from a mixer channel with separate level control. Usually used to create monitor mixes or as effects sends. See also: Prefader; Postfader.

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

Bodypack – A receiver style which can be worn on the body.

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

Comb Filtering – The variations in frequency response caused when a single sound source travels multiple paths to the listener's ear, causing a "hollow" sound quality. The resultant frequency response graph resembles a comb.

Compressor – A signal processor that reduces the level of incoming audio signals as they exceed a given threshold. The amount of reduction is usually defined by the user.

Equalizer – A signal processor that allows the user to boost or cut selected frequencies. Used for tone shaping and limited feedback control. Variations include graphic or parametric.

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

Fidelity – A subjective term that refers to perceived sound quality.

Frequency Agile – Having the ability to change frequency: tuneable.

Frequency Response – Variation in amplitude of a signal over a range of frequencies. A frequency response graph is a plot of electrical output (in decibels) vs. frequency (in Hertz).

Front-of-House (FOH) – Usually refers to the mix created for the audience, or "house."

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

Gate (Noise Gate) – A signal processor that mutes the audio signal when it drops below a given threshold.

Ground Plane – Electrical approximation of a zero-potential reflective surface at the base of an antenna.

Insert – A routing point on a mixer that allows an audio signal to be sent to an external device and returned back to the same mixer channel.

Isolation – Freedom from leakage; ability to reject unwanted sounds.

Limiter – A signal processor that prevents signal levels from exceeding a certain threshold.

Mix – A combination of input signals that are varied in level and tone to provide the desired sound for the listener.

Mixer – A device which allows the combination, manipulation, and routing of various audio input signals.

Mono – A single channel of audio.

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

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

Operating Frequency – The final output frequency of a transmitter or the tuned frequency of a receiver.

PA – Public Address. Usually refers to a sound system.

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

Pan – The ability to move a signal across the stereo field, from left to right, or vice versa.

Prefader – Auxiliary sends whose output levels are unaffected by the main mixer channel level control.

Postfader – Auxiliary sends whose output levels are directly affected by the output level of the main mixer channel level control.

RF – Radio frequency.

Receiver – Device which is sensitive to radio signals and recovers information from them.

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.

SPL – Sound Pressure Level is the loudness of sound relative to a reference level of 0.0002 microbars. Usually expressed in dB SPL.

Sound Reinforcement – Amplification of live sound sources.

Stereo – Two channels of audio, left and right, which can be used for panning audio signals to simulate a realistic listening environment.

Subgroups – A way of combining audio signals from individual mixer input channels into smaller groups.

Supercardioid Microphone – A unidirectional microphone with tighter front pickup angle (115 deg.) than a cardioid, but with some rear pickup. Angle of best rejection is 126 deg. from the front of the microphone.

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

Transmitter – Device which converts information to a radio signal.

UHF – Ultra High Frequency (about 300-1000 MHz).

VHF – Very High Frequency (about 30-300 MHz).

BIOGRAPHY: Gino Sigismondi

Gino Sigismondi, a Chicago native and Shure Associate since 1997, has been active in the music and audio industry for nearly ten years. In addition to his work as a live sound and recording engineer, Gino's experience also includes performing and composing. Gino earned his BS degree in Music Business from Elmhurst College, where he was a member of the Jazz Band, as both guitar player and sound technician. After graduation, he spent several years working for Chicago area sound companies and night clubs, before settling down to a select group of the area's top local acts. As a member of Applications Engineering, Gino brings his years of practical experience to the product training seminars he conducts for Shure customers, dealers, distribution centers, and internal staff. He has also authored several Shure applications bulletins as well as written for the Shure Web site. Gino continues to remain active as a sound engineer, expanding his horizons beyond live music to include sound design for modern dance and church sound.

Special thanks to these Shure Associates who worked hard to get this project finished, and/or provided invaluable technical assistance:

Davida Rochman
Scott Snyder
Jon Halverson
Chris Zachara
Tim Vear
(for letting me steal from his Wireless book!)

Thanks also go to:

Thom Fiegle (Sensaphonics)
Michael Santucci (Sensaphonics)
LargerPond Marketing

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LISTEN SAFE

Shure Listen Safe is the Company's hearing conservation program, which is dedicated to educating musicians, audio professionals, and consumers on how to enjoy sound in a safe, responsible way. Donated funds are used to underwrite the cost of hearing screenings, creating educational materials, and funding hearing research studies. In addition to these donations, Shure's activities include providing free hearing screenings and distributing hearing protection devices at pro audio industry trade shows, music conferences and festivals, and to its employees. For more information about Shure Listen Safe, please visit www.shure.com/hearing.

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AL1456G 5M 6/09