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INTRODUCTION

HOME RECORDING and PODCASTING

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Home Recording

Introduction to HOME RECORDING AND PODCASTING

Introduction

Home recording and podcasting are increasing in popularity every day. The equipment used for these applications has become more sophisticated, practical, accessible, and affordable — and more and more people are getting involved with these types of audio projects.

Shure has been a leader in audio production for more than 80 years, and we have learned that “audio quality” will make or break any project.

The purpose of this guide is to help you, the home recordist or podcaster, capture better sound for your recording projects, whether they are ...

- *monologues,*
- *round table discussions,*
- *interviews,*
- *music performances, or*
- *creating audio tracks for videos.*

If you are new to home recording and podcasting, this is a great place to start!

If you have been wondering why other people are able to create better sounding recordings, this guide should help you identify the reason and the solution.

The guide takes a step-by-step approach to discussing principles, products and placements, as well as helping you get past some of the most common problems. We've also provided reference sources where you can find more technical information.

So put down your iPod (just for a few minutes) and learn why some podcasts and home recordings are clean and crisp and why others sound like they were recorded from the bottom of a can or in front of an air conditioning unit.

Introduction



SECTION ONE

GETTING STARTED

The most important concept of all: Start with Good Sound.

Good sound starts with one basic principle: The better the original source recording, the better the final audio output will be.

Therefore, your #1 goal is to make sure that your initial recording of voices and instruments provide good starting points for mixing and combining into a final program.

To give yourself the most flexibility, you need to be able to control and edit each voice or instrument individually.

In professional recording studios, every individual voice, instrument, and sound effect is recorded separately, so that they can be blended together in just the right mix. In a home studio, you might not have the time or equipment to record every single thing separately, but it's a good idea to at least separate voice tracks from instrument tracks, or separate different types of instruments from each other (one track for drums and one for guitar, for example.) This will allow you to adjust the level or tonal quality of each voice and instrument separately, so that the changes you make to one track are not also applied to another track that does not need it.

If you are setting up a home recording studio or are just about to begin your first podcast, you need to keep this guiding principle in mind during all three of the following major steps:

- choosing your recording location
- selecting your equipment, accessories, and even cables and adapters
- recording the audio

While you can clean up the sound, somewhat, during the editing and mixing process, the equipment quickly becomes expensive and the techniques start to get very advanced. Even if money and expertise are at your fingertips, there is only so much that even the world's best sound engineer can do to soften the negative affects of background noise and hiss without noticeably altering the sound of voices or instruments.

Simply put: the best way to eliminate unwanted and ambient noise is to make sure you do not capture it in the first place.

The difference between home recording and podcasting.

Home recording is the term used for any recording created outside of a traditional professional recording studio. This term can be misleading, however, because recording equipment has become so compact, easy to use, and affordable that there might be no difference between the equipment used in a traditional recording studio and what is used in a home 'project' studio. Today, a recording studio might be located in an office, basement, or garage and be capable of producing professional-quality content. A traditional recording studio still offers one critical component that is missing from most home recording studios: a quiet room that is suitable for recording sound.

Introduction to HOME RECORDING AND PODCASTING

The word *podcasting* comes from combining ‘broadcasting’ with Apple’s popular audio player, the iPod. It is now the generic term for creating audio or video files that can be subscribed to, so that the subscriber is automatically notified that a new episode is available. It is also a generic term for downloadable (non-subscriber) recordings – especially those that are part of an ongoing series.

Podcasts can contain audio only, audio with graphics, or video, and can be played on an MP3 player or a computer. Some podcasts are created by traditional media organizations (news networks, magazines, newspapers, etc.) while others are created by companies, religious groups, educational institutions, or individuals. The one thing they have in common is the desire to deliver information to people who are interested in a particular subject.

The bottom line is that *home recording* is a method of capturing and producing content, while *podcasting* is a means of distributing that content. In other words, many people reading this booklet are both home recordists *and* podcasters.

So, What *is* Good Sound?

Subjective as the term ‘good sound’ might seem, it’s the optimization of three components:

1. Audibility – is it *loud* enough?

More to the point, does the sound achieve sufficient level that does not require the listener to strain to hear it or reach for the volume control?

2. Intelligibility – is it *clear* enough?

Intelligibility describes how well the listener can understand the information being delivered. Intelligibility is critical in spoken word recordings, because it determines whether the listener accurately hears the difference between words like “cat” and “bat”. Poor intelligibility can be the result of poor diction or pronunciation by the talker, poor recording, or simply too much undesired ambient sound or noise.

3. Fidelity – does it sound like ‘being there’?

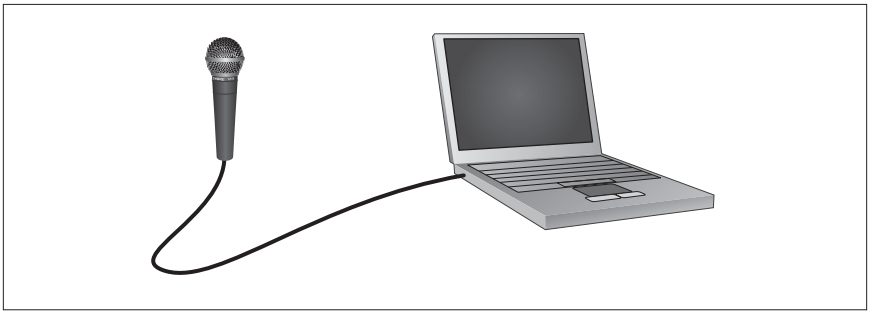
Each component of the sound path can affect the tonal character of the sound arriving at the listener’s ear. This changes the realism and accuracy of the recording. While the listener can hear your words and music clearly and at a comfortable level, if it does not sound like the original performance, then it lacks fidelity.

You may hear many colorful, yet vague, terms for describing sound – such as ‘warmth,’ ‘punch,’ and ‘bottom’ – but they’re not measurable. For example: Speech that is warm but unintelligible does the listener little good.

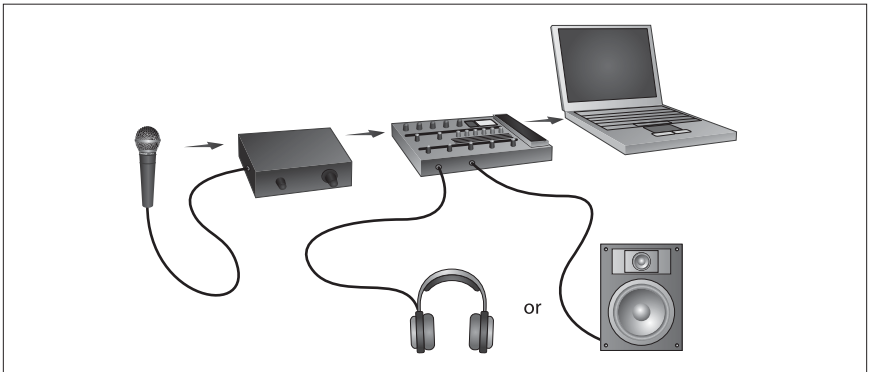
But, at the end of the day, “good sound” is 100% subjective. If you and your listeners *believe* your recordings and podcasts have good sound (that is, they sound like what you have in mind and people can understand them) then you *have* achieved good sound. Now comes the hard part: getting your recordings to sound as you intended.

Basic equipment needs

Below are basic and more complex recording signal paths for home recording and podcasting needs.



Basic: Mic → Cable → Recording Device (often the computer)



More complex:
Mic → Cable → Preamplifier →
Mixer/Signal Processor + Loudspeaker or headphones → Recording Device

Before you create a budget and run out to buy all the hardware you can afford, it’s critical to match your equipment to your recording environment.

SECTION TWO

CAPTURING SOUND

The Recording Environment

The 'deader' the better.

As recording equipment becomes more compact, you can look to smaller and smaller spaces for your home recording and podcasting needs. This is good because it provides more opportunity to find the most acoustically neutral (quietest and least reflective or 'dead') area. Smaller areas also make it easier and less expensive to improve the acoustics.

So what are you looking for? It's more what you are looking to *avoid*:

- *Avoid* reflective or hard surfaces, such as windows or concrete walls. If your space includes these surfaces, you will want to 'deaden' them with acoustic foam, carpet, blankets, or heavy drapes.
- *Avoid* fans, air conditioning units, furnaces or other appliances that generate mechanical or electrical noise.
- *Avoid* other people. Use a sign to alert family and friends to stay away while you are recording.

Make a few sample recordings.

Your brain is quite good at ignoring background hiss or mechanical noise, so make a recording of the chosen areas for a minute or so. Then listen to them on your MP3 player or computer. You'll be surprised at how much noise the microphone picks up that you didn't notice while standing in the room. Obviously, the area with the least amount of noise should be the winner.

Tip: *Can't find a "dead" enough space? Sing or talk into the closet.*

One of the best tricks we've found is to open the closet door, throw blankets or sleeping bags over the wide open doors, and position the mic so that the least sensitive side of the mic is facing into the back of the closet. The more clothes in the closet, the better! This absorbs many of the reflections and can give you a more intelligible sound.

Microphone styles

The choice of microphone style is determined by what the sound source is (a person vs. an instrument), what the sound source is doing (sitting, standing, moving, not moving, etc.), and how the microphone can be conveniently mounted. The microphone style should help put the microphone in exactly the right place *and keep it there*.

Handheld – True to its name, the handheld microphone is shaped to be easily held in the hand. Handhelds can, of course, be attached to a mic stand to free up the hands. Mic stands are a good idea for home recording. One of the main variables to improving sound levels and overall sound quality is to keep a consistent distance between the sound source and the microphone. Placing the microphone on a stand also reduces *handling noise*, which are low frequency ‘thumps’ created by the talker’s hand tapping or interacting with the body of the microphone.



Handheld
microphone



Stand mounted
microphone

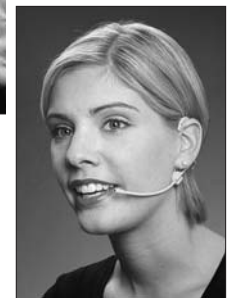
Stand mounted – These mics are designed to be attached to a stand and, for this reason, additional accessories, such as shock mounts, work best with these microphone types.

Lavalier – Very small microphones that clip to a shirt or tie, lavalier microphones are good for people who need to move while talking (examples: presenter, theatre performer) or who may be intimidated by seeing a microphone in front of them (such as an interview subject for a podcast). Lavalier microphones also free up the hands for other uses, such as demonstrating how to operate a tool or perform an activity.



Lavalier
microphone

Headworn – These mics take the lavalier to the next level. While the lavalier is fixed in position on the body, a headworn mic is fixed in relationship to the mouth, keeping the distance between the sound source (the person’s voice) and the microphone constant. Because the microphone is closer, there is less pickup of background noise in a noisy recording environment.



Headworn microphone

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Instrument-mounted – Some microphones are specifically designed to attach to certain instruments, such as the bell of a trumpet or the edge of a snare drum. This helps isolate the sound of that particular instrument from others around it, while keeping the distance from the microphone to the sound source constant. (See example to the right.)

Direct input – Some instruments, such as an electric bass guitar, can be connected directly into a device called a *direct box*, which converts the instrument signal to a standard microphone signal. The direct box is not a microphone, but it replaces the microphone in the signal path. The advantage of using a direct box is that you eliminate the microphone, so there is no ambient noise in the recording. The disadvantage is that some instruments may not sound as expected when recorded directly, because you are missing the tonal character that is added by the body of the instrument (or, in the case of an electric guitar or bass, by the guitar amplifier).



Instrument
mounted
microphone



Example of a direct input
(or “direct box”).

Tip: If you use a direct input, you should also consider using a monitor speaker so that you can hear exactly what you are recording. (See “Monitors” on page 24.)

Transducer Types

The *transducer* is the part of the microphone that converts acoustic energy (sound) into an electrical signal that can be recorded. There are several types of transducers, but the two most common types used for recording are Dynamic and Condenser.

Dynamic mics are very rugged and can handle high sound pressure levels, like those delivered from snare drums and guitar amps. They are also very tolerant of hostile conditions like high temperature and humidity... and careless interview subjects.

Condenser mics are more sensitive and more responsive to higher frequencies, which allows them to capture more detail from instruments like acoustic guitars or cymbals. Condenser microphones can also reveal added texture and realism in voices.

There are two main types of condenser microphones:

Small diaphragm – generally used for live performance and recording. They are called *small diaphragm* because the transducer’s diaphragm is less than one inch in diameter. (See diagram below.)

Large diaphragm – traditionally favored by recording studio engineers and broadcast announcers, condenser microphones with a large diaphragm (one inch in diameter or larger) usually have higher output, less self-noise (the ‘hiss’ the microphone might make), and better low-frequency response, which can result in a ‘higher fidelity’ sound for both vocals and instruments.



Does Diaphragm Size Really Matter?

When looking for a large diaphragm microphone, you will come across terms such as ‘large style’ and ‘large diaphragm type.’ Generally these do not refer to true large diaphragm microphones, but to smaller diaphragm microphones that are designed to appear as if they were large diaphragm versions. A true large diaphragm microphone typically has a physical diaphragm of 1” or larger diameter. (As shown in the illustration above.)

But the question remains... “Does it matter?”

As costs for some true large diaphragm microphones are now competitive with small diaphragm microphones and even dynamic microphones, it becomes a question of performance and personal taste.

The answer is... it depends. If you want the highest sensitivity, the lowest self-noise, and a truer sound, go for a large diaphragm mic. In all cases... try them before you buy them. But recall that ‘good sound’ is subjective. If you like the way your voice sounds in your recordings, regardless of diaphragm size, then you have found the right microphone.

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USB microphones

More and more professional microphone companies are manufacturing USB microphones. A USB microphone is essentially a microphone that has a USB audio interface built in, so that it can be plugged into your computer without requiring an external audio interface.

In addition, some USB microphones provide a headphone output to help you monitor your recordings. (See “Monitors” on page 24.)



XLR connector

USB connector

To USB or not To USB

If you only need to use one microphone and plan to record directly into your computer, a USB microphone is the most convenient solution. It's quick and easy to set up, and it eliminates the need for a pre-amp, mixer, or USB audio interface.

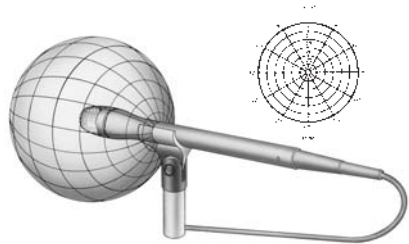
If you need to use multiple microphones at once (to record multiple voices or instruments, for example) or you want to be able to connect the microphone to a mixer or other recording equipment as well as your computer, consider an XLR-to-USB adapter. This lets you connect existing XLR-style microphones into the USB port on your computer.

Pickup patterns

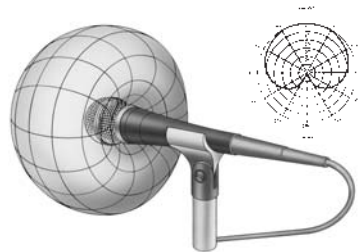
Microphones are available with various pickup (or polar) patterns. The pickup pattern is the representation of the microphone's directionality. In other words, the pickup pattern describes the microphone's sensitivity to sounds arriving from different directions.

Omnidirectional – picks up sounds equally from all directions. Good for natural room sound and group vocals. Also good for when the singer or talker may move around different sides of the microphone (but their distance to the mic stays the same). However, if the goal is to eliminate ambient sound, a unidirectional pickup pattern is a better choice.

Unidirectional – most sensitive to sounds coming in front of the mic, and less sensitive to sounds coming from the sides or rear. Unidirectional mics fall into two main categories:

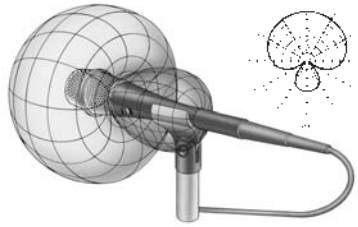


Omnidirectional



Cardioid (Unidirectional)

Cardioid – the most common type of microphone. Called ‘cardioid’ due to its heart-shaped pick up pattern (see diagram on page 12), this microphone helps reduce pickup of background noise or bleed from nearby sound sources. However, if the goal is to enable the listener to hear what is occurring in the background, you should consider an omnidirectional pick-up pattern.



Supercardioid

Supercardioid – is even more directional than the cardioid. Supercardioids have the tightest pickup pattern, further isolating the sound source. Good for noisy, crowded spaces and when multiple microphones are being used, such as for round-table discussions where you want to keep the voices distinct.

An example: Annie's knitting podcast.

Annie hosts a knitting show. If she wants the listener to hear the ‘click and clack’ or her darning needles during the podcast, she should use a mic with an omnidirectional pickup pattern. If not, she should use a mic with a cardioid pick-up pattern.

Better yet, she can use two mics. A cardioid aimed at her mouth, to pick up her explanations, and another cardioid aimed at her hands, to pick up the click and clack of her needles. Then she can mix them together – adjusting the levels of the two sounds to achieve the maximum combination of intelligibility and fidelity – during the editing process.

Microphone electrical outputs

The electrical characteristics of the microphone's output signal are important, because they must be compatible with the audio input to which the microphone is connected. Microphone signals have historically been *analog*, but with the increasing popularity of *digital* audio recording and editing, microphones with digital audio output have become available.

Some ‘purists’ prefer analog, while others prefer having the microphone provide the digital signal. The truth is that you need to end up with digital, so it really depends on how *and when* you want to enhance your sound. (See Appendix for a more complete discussion on analog and digital output.)

Phantom Power and Condenser Microphones.

Condenser microphones require power. A common way of providing them with power is through something called *phantom power*: Phantom is real power, but is the term used for when a voltage is fed through the microphone cable. Most microphone mixers provide phantom power, so if you are using a condenser microphone and a mixer, make sure the mixer has a microphone input that provides phantom power.

Note: Dynamic microphones do not require phantom power, nor will they be harmed if they are plugged into a microphone input that has phantom power turned on.

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Microphone Accessories

Mounting accessories can help improve the resulting sound of your home recordings and podcasts. Creative use of these accessories can allow microphones to be placed almost anywhere, either freeing up your hands for other needs or helping make sure the microphone is in precisely the right place.

Stands, Booms and Goosenecks—should be sturdy enough to support the microphone in the intended location and to accommodate the desired range of motion. These accessories come in many shapes and sizes, but the purpose remains the same: to position the microphone in the right place to pick up only the sound you want. So finding the right version for your needs and your microphone is what matters most.

Shock Mounts—are used to isolate the microphone from vibrations transmitted through the stand or the mounting surface, such as a desktop or floor. A shock mount can reduce or eliminate the ‘handling’ noise you hear when microphones are moved during a recording session or if the surface upon which the microphone rests is being jarred or vibrated (often called ‘stand thumps’).

Windscreens and Pop Filters—First of all, you need to understand what *popping* is. When you say the word ‘pop’ for example, you will hear an explosive breath after the ‘p’, that is, ‘po-puh’. Pops occur most often with “p”, “t”, “d”, and “b” sounds, and can be very distracting on the final recording. Windscreens and pop filters provide an acoustically transparent shield around your microphone, which breaks up the wall of air before it hits the mic and helps reduce popping sounds. (See Technique 2 and diagram on page 17.)



A boom arm stand, while more expensive than a normal stand, balances the microphone in mid-air and can be repositioned with gentle pressure that, if careful, will not be heard in the recording.



Example of a shock mount and a pop filter.

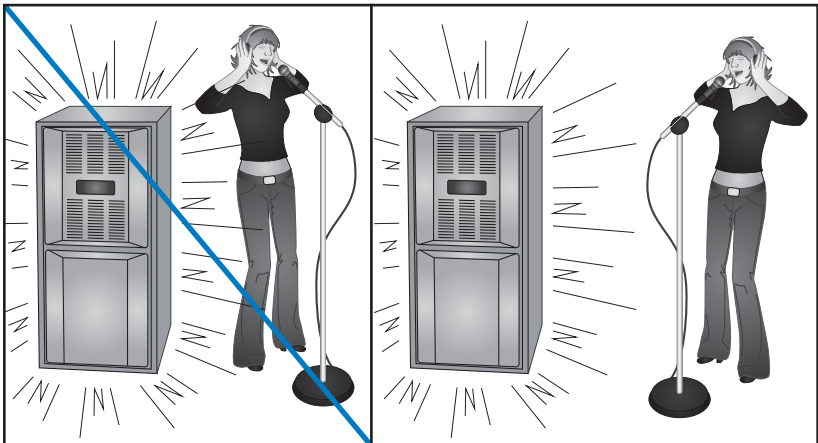
Two Microphone Accessory Tips:

- Too much handling noise? – Or does it sound like the microphone is being tapped during the recording (stand thumps)? Use an accessory shock mount.
- ‘P’, ‘t’, ‘d’, and ‘b’ sounds too explosive for your tastes? Try a ‘pop’ filter. **Note:** You can also place the microphone out of the path of pop travel, such as a few inches to the side, above, or below the mouth. (See diagram on page 17.)

General Recording Techniques

Within this guide, we will hit the highlights applicable for general home recording and podcasting needs.

- 1. Microphone technique is largely a matter of personal taste.** Whatever method sounds right to you *is* right. Keep trying different placements until you get the sound you want.
- 2. Maintaining a consistent sound level is critical.** Make sure your sound sources do not move in and out of the pickup areas of the microphone. Movement (i.e. varying distance from the microphone) will change recording levels, which are hard to fix later.
- 3. Keep the microphone away from reflective surfaces.** Reflections (i.e. bouncing sound waves) caused by hard surfaces, including even tabletops and music stands, can affect the sound quality captured by the microphone. This is also called reverberation and if you want this effect, it is best to add it later. (See “Reverberation” on page 24.)
- 4. Place the microphones far from unwanted sound sources.** Be sure to also point the microphones away from any unwanted noise. Make a few sample recordings of the chosen recording area, with the microphone facing different directions, to find the quietest possible placement.

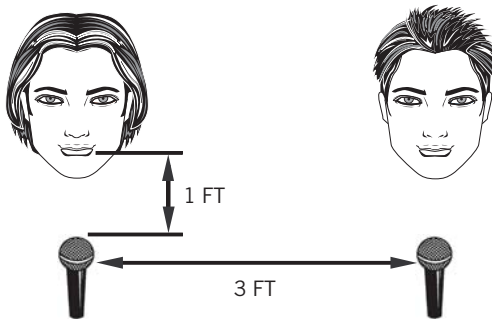


Picking up unwanted background noise (such as from street noise through windows or from mechanical appliances including furnaces, heating and A/C registers) is a common issue with home recordings.

The more you can isolate your recordings from background noise, the better they will sound.

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- 5. Make sure to use a microphone with the right pick-up pattern and transducer type for the application.** (See “Microphones” on page 9).
- 6. Use as few microphones as necessary.** Fewer microphones mean fewer technical issues and, for the purposes of capturing clean sound, less pickup of background noise.
- 7. Record each voice and instrument (or sound source) separately.** (See “Start with Good Sound” on page 5.)
- 8. Keep the 3-to-1 rule in mind.** When multiple microphones are used, the distance between microphones should be at least three times the distance from each microphone to its intended sound source. For example, if two microphones are each placed one foot from their sound sources, the distance between the microphones should be at least three feet. If each microphone is just two inches from an instrument, they only need to be six inches apart.

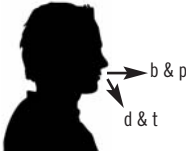
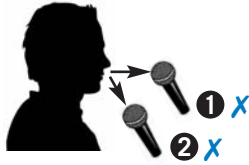

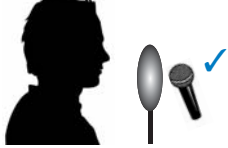


The 3 to 1 Rule: The distance between microphones should be at least three times the distance from each microphone to its intended sound source.

Vocal Microphone Placement Techniques

In addition to the general techniques, here are a few specific to vocal recording:

- 1. Keep the microphone 6" – 12" from your mouth.** This distance is close enough to minimize pickup of unwanted room reflections and reverberation, but far enough away to minimize picking up mouth and breathing noises. Do not get too close, though. 'Eating the microphone' can decrease intelligibility.
- 2. Aim the microphone toward your mouth from below or above.** This placement minimizes the popping caused by plosive consonants (e.g. "p" and "b"; "d" and "t"). Every person is different, so some trial and error is usually necessary. Some people have problems with "p" but not "t", for example.
- 3. Use an external pop filter.** Though most microphones have some sort of built-in windscreen, an additional filter will provide extra insurance against "p" pops. The pop filter also helps serve as a reference to help you maintain a consistent distance from the microphone.
- 4. Speak directly into the microphone.** When you turn your head away from the microphone, the sound captured by the microphone will get noticeably duller. If a talker continues to move away from the mic, try a headworn microphone. (See page 9.)

Plosive Directions	Risky Microphone Placements
 <p>"b" & "p" tend to go straight out. "d" & "t" tend to drop.</p>	 <p>① "b" & "p" aims at mic causing popping. ② "d" & "t" aims at mic causing popping.</p>
Suggested Microphone Placements	Use a Shield
 <p>③ Place mic <u>above</u> plosives to avoid popping. ④ Place mic <u>between</u> plosives to avoid popping.</p>	 <p>A pop filter can shield the microphone from plosives. (See page 14.)</p>

Plosives (popping) and tips for avoiding them with microphone placement

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Instrument Microphone Placement Techniques

When recording instruments, it's always best to find placement techniques specific to the instrument you are using. Plus, the same as for vocal recording, it's largely a matter of personal taste. Whatever sounds right to you *is* right.

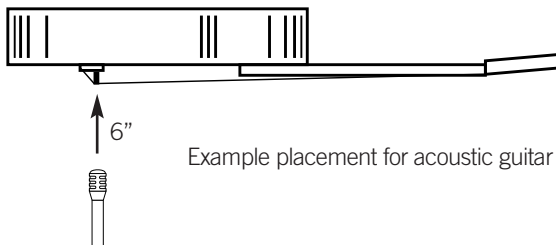
However, for the purposes of this guide, we will provide a few pointers:

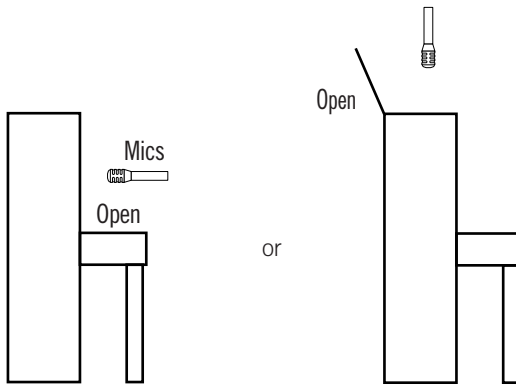
- 1. Try a few different recordings and see which placement sounds best to your ears.**
- 2. Record any instruments separately from vocals.** Use two microphones: one positioned near the singer's mouth and one positioned near the instrument's sound source.
- 3. Place the microphone only as close as necessary.** Too close a placement can color the sound source's tone quality (timbre), by picking up only one part of the instrument. But too far away means you will pick up more ambient sound.

Tip: To determine a good starting microphone position, try closing one ear with your finger. Listen to the sound source with the other ear and move around until what you hear sounds good. Put the microphone there.

Different instruments require different microphone placements and techniques. We can highlight some of these differences by looking at a couple of specific instruments:

Acoustic guitar – Assuming you do not have multiple microphones, you can achieve a well-balanced, natural sound by placing the microphone six inches (6") above the side, over the bridge, and in line with the front soundboard (see graphic). Some people prefer clipping a miniature microphone outside of the sound hole, which allows freedom of movement.





Upright Piano – Assuming you only have one instrument microphone, try placing it just over the open top, above the treble strings. If you can remove the front panel, try aiming it at the front hammers.

Piano – Open the lid and aim a condenser microphone just over the top, above the treble strings.

Tip: Minimize pickup of floor vibrations by using a rubber *shock-mount* for the mic. (See “Accessories” on page 14.)

Again, there is no *right* solution, there are only *techniques* that provide a good starting point for your own experimentation. (See “Additional Resources” on page 27 for places where you can get specific advice for your instrument recording needs.)

SECTION THREE

ENHANCING

Recording Devices

The end point of the signal path (before sharing) is the recording device. In many home recording studios, the recording device will be the computer. (See “Basic Equipment Needs” on page 7.) However, some audio projects require more complex equipment needs or require sounds recorded outside of the studio. In any case, let’s look at four common recording devices.

Digital recorder – can be a standalone unit or part of a ‘studio-in-a-box’ solution. Some features of digital recording devices include simple editing and instant CD burning. Some allow you to create both MP3 and WAV files and even convert files between the two for easier file distribution.

There are also inexpensive and easy-to-use portable voice recorders for use in the field (outside of the studio) for capturing interviews or other non-music recordings. For some podcasters, this one device could be all the recorder you need.

Analog recorder – often associated with old-fashioned reel-to-reel systems but can still be found in the studios of musicians looking for a certain sound.

Sound card – is the hardware circuit board inside your computer. If your computer has sound, there is a sound card in there. Usually it’s plenty for your video games and office work, but is it powerful enough for your recording needs? If not, consider a...

Audio interface – is an external device that sits between your audio inputs/outputs and your computer. Think of it as an external sound card. An audio interface can have either a USB or Firewire port to connect to the computer, although USB ports are more common on most computers.

Mixers/Interfaces

At the nerve center of the prototypical professional recording studio is a giant mixing board. Odds are, you have a picture in mind from a movie you have seen about an old-time rock band.

Today, especially for home recording needs, you can find far more compact mixers and, in many cases, you might be able to go ‘mixerless’.



Example of a portable digital recorder.
*Image of PMD620 courtesy of
Marantz Professional.*



Example of an audio interface box.

Going 'mixerless' all depends on ...

1. How many audio inputs you are combining together
2. Whether your interface (software or audio sequencer) provides the control you need
3. Whether or not you have lots of external hardware (such as synthesizers, reverbs, delays, pedals, etc.)
4. If you plan to produce CDs or DVDs, versus only podcasts or digital recordings
5. Your budget

Do you need a mixer or an interface? Or: Just how fast is your computer?

The mixer, itself, does not define your home recording studio. While a mixer does provide that professional studio look, it might be entirely unnecessary for your needs. In fact, more and more professional sound studios are going 'mixerless.'

This does not mean they are giving up the ability to control multiple audio tracks. It means that are relying on audio interfaces and their computer to do the mixing.

The good news, here, is that going 'mixerless' often costs less and provides a more portable/moveable studio.

However, the less powerful your computer's sound card, the more likely you will need an external interface or, even, a mixer if you plan to do any serious audio manipulation.

Also, while some software (such as Apple's GarageBand) does not require an interface, other software (such as Digidesign's Pro Tools) does require a compatible audio interface to be connected for the software to work.

In addition, plugging your microphone directly into your computer's sound card might not provide the sound quality you want. Many internal sound cards are not shielded from electrical noise that is often caused by fans, hard drives, and the computer's own circuitry. Because of this, noise and hum can be introduced into your audio. Also, most internal sound cards are not equipped with professional microphone connectors, do not provide phantom power for condenser microphones, and do not provide enough amplification or *gain* when working with low-level tracks or sounds.

So... What's our advice? Try using your computer. If you are happy with the resulting sound, then you do not need a mixer.

Introduction to HOME RECORDING AND PODCASTING

Mixers

Some recording devices do not allow you to connect a microphone directly, though this is becoming less and less common. More likely, there will be cases when you are capturing a number of audio sources and will want to manipulate and combine these tracks into a single recording, which is called a *mix*.

In both these cases, you will want to consider a mixer (or ‘mixing console’ or ‘mixing board’). Mixers come in many varieties – and can get expensive and technical very quickly – so make sure you only get as much mixer as you need.

Be sure that the mixer you choose has enough inputs to handle the number of microphones you are using. You also need to consider what you might need a few years from now. You can’t easily add more inputs, so if you do invest in a mixer, leave a little room for growth. Also, if you wish to capture the sound of each microphone on a separate track (which we wholeheartedly advise), then your mixer will need to have direct outputs for each channel to connect to your recording device. Otherwise, you will only be able to capture the blend of all of the inputs.

Interfaces

An audio interface is what gets the sound into and out of your computer. Technically, it converts an *analog signal* from microphones and other audio equipment to *digital data* (so you can manipulate and share it).

There are two basic types: a sound card (usually built right into your computer) and a standalone audio interface box. (See example on page 20.)

Microphone Pre-Amp

A good microphone pre-amplifier (called a ‘preamp’ or ‘mic-pre’) boosts the sound level from your microphone without adding noticeable hiss, and often does a better job of rejecting electrical noise and hum than a mixer or audio interface. A mic preamp usually accommodates just one or two microphones and offers separate controls for each.

The microphone preamplifiers built into most USB or Firewire audio interfaces are adequate for the vast majority of home recording applications. But if you are recording a quiet voice or instrument, placing the mic far from the source, or just using a microphone that is not very sensitive, you might find that your mixer or audio interface just can’t amplify the signal enough without adding excessive hiss. If this is the case, you should consider using an external preamp to increase the sound strength to a more audible level. Additionally, some recording interfaces may have as many as eight inputs, but often only two will accept microphone level signals. An external pre-amp lets you take advantage of these additional inputs.



Example of a microphone pre-amp.

Cables and Adapters

While it's best to try to find new equipment that is compatible with your current equipment, using adapters is not as bad as you might think. Six adapters in a row to connect your mixer to your recording device? Perhaps that's too many and you might hear some noise or hum. But don't replace the equipment if all you need is an adapter. Odds are, any hiss or issues are not coming from the adapter.

Better advice, however, is not to skimp on the cables. Finding a quality cable, of the right type for your equipment, is important for reducing noise and hum.

Recording Software

Now that you have your sound captured, you need to edit, mix and apply effects. Some software also includes extensive sample libraries (samples of real instruments that can be played back or 'triggered' by a keyboard, for example) and synthesizers which mimic various instruments.

Recording software is all about mixing (which was covered earlier), effects processing (which you can read about below), and which export features you want (See "Saving and Sharing" on pages 26-27.) Make a list of what you are planning on doing and make sure the software you choose has all that functionality. Also, some software requires an interface (See "Mixers/Interfaces" on page 21). This is good to know before you open the box.

Effects Processing

Effects processing is a means of manipulating or changing certain aspects of the recorded sound with the goal of enhancing the overall sound quality. Sometimes an effect is applied to an individual track (vocal, guitar, etc.), while other times it is applied to the finished recording.

There are plenty of effects you can use for home recording, though most podcasts (especially if you capture clean sound) do not require much in the way of effects processing. While you can find a separate piece of hardware to provide each unique effect, most are commonly included as unique controls within a single software package or digital mixing console.

Compression – A compressor automatically turns down the talker's or singer's peaks (loud parts) by a preset amount so they don't cause distortion. Compression also reduces the difference between the loudest and softest note, so the apparent loudness is greater. Example usage: A singer might vary in loudness from very soft to very loud, but the compressor reduces the magnitude of these extreme changes.

Limiting – Limiters are like compressors, but instead of *reducing* any levels that go beyond the preset threshold, *they stop* them from getting any louder at all, that is, providing a top limit that cannot be exceeded. How is this different? Imagine a sound that goes to 11. The compressor might reduce it to 10 or 10.5 depending on the dynamics of the signal, but the limiter would cap the sound firmly at 10 no matter how abrupt that might sound. Note that compressors and limiters affect all voices or instruments on the same track equally. If one vocalist yells into their mic causing the compressor to reduce the level, other voices on the same track will be similarly affected.

Introduction to HOME RECORDING AND PODCASTING

Equalization (also called EQ) – EQ emphasizes or de-emphasizes certain frequency bands, which can either make different tracks stand out from each other or helps different-sounding tracks sound more similar. Equalization is critical when combining multiple instruments into a single mix.

Advanced Tip: Avoid the dreaded ‘smiley-face’ EQ curve.

As a general rule, you should try to shape the sound by reducing certain frequencies, rather than boosting others. In particular, excessive boosting of low frequencies is a common cause of less intelligible recordings. These ‘muddy-sounding’ speech recordings often result from the dreaded ‘smiley-face’ EQ curve, when lows and highs are boosted to the point where the all-important mid-range (critical to intelligibility) is effectively masked.

Reverberation (or ‘*reverb*’) – Reverb is the ‘bouncing around’ of sound waves in a particular acoustic space, such as a room. (See “General Recording Techniques”, tip 3, on page 15.) Reverb is often added to audio tracks to create a feeling that the recording took place in a particular environment instead of in a recording studio. Example: In a movie, you might want the talker to sound like she is in a subway station. Adding the right amount of reverb will allow you to achieve this effect.

Delay – Delay is adding intentional latency to an audio signal, which is usually combined with the original undelayed signal for creative effect. Like reverb, delay can be used to mimic the sound of a particular environment. (See Appendix for a more complete description of latency.)

Normalization – Normalization adjusts the levels of different audio files or parts of a file to be the same. Audio tracks that come from different sources or that have been recorded at different times can have different levels. Since these level differences might be noticeable to listeners, you can normalize them to make it sound more like they were recorded together and so it is less of an audio rollercoaster for anyone enjoying your podcast.

Monitors

Why Monitor Your Recordings?

Before you can record or mix good sound, you need to be able to hear what you are getting. In audio terms, monitors allow you to listen to the audio while it is being recorded or edited.

- If you are mixing or editing different sounds that were previously recorded, you can monitor through loudspeakers, headphones, or earphones.

- If you're trying to sing or play along with a recorded music track, you need to monitor through headphones or earphones. If you try to do this with speakers, the sound of the speakers will be picked up by the microphone, which could in turn 'bleed' onto the new track, or be reamplified through the speakers. This can cause the whistling or howling you know as *feedback*.
- If you are trying to sing or play along in sync with previously-recorded tracks, you must be able to monitor in real time, with almost no latency. (See "Latency" in the Appendix.) A few milliseconds of latency isn't critical, although it can slightly alter your perception of pitch or tone. More than 10 milliseconds of latency can have a noticeable effect on your rhythm and timing.

To monitor in real time, you need to tap into the audio signal *before* it gets converted from analog to digital and fed into the computer.

Tip: Some USB microphones and audio interfaces provide a headphone output, which allows you to monitor without latency.

The Basic Types of Monitors

Speakers – Many home recordists start with their handy stereo system, which already has speakers attached. As you move into more sophisticated recording, you should consider a pair of dedicated 'studio monitors.' The most common choice for home recording are called 'near field' or 'close field.' These are designed to provide accurate sound when you are very close to them, as opposed to normal stereo speakers, which are designed to sound better from a distance and are not necessarily accurate.

Headphones or earphones – The simplest and easiest to understand monitor is a set of headphones. Earphones (also called in-ear earphones) are basically headphones that are inserted directly into the ear. Either of these will let you hear sounds without outside distractions and without bothering other people around you.



Types of monitors (left to right) speaker, headphones, and earphones.

SECTION FOUR

SAVING & SHARING

Now that you have your recording, two questions arise:

1. How are you going to save it?
2. How are you going to get it to your audience?

The easiest way to save your files is on your computer, but anyone who has experienced a computer crash knows that this is not optimal. Also, these files can get really big really fast. Most home recordists and podcasters have a number of backup solutions for storing their recordings:

- **Backup to an external hard drive** – dedicated to storing your recording files. In this case, the biggest (in GB) and the fastest (in RPM) you can afford the better.
- **Burn onto CDs/DVDs** –These are being used mainly as a backup to your external hard drive... but *do* make sure to make new versions periodically. The good news is that blank CDs and DVDs are fairly inexpensive.

Tip: When saving your files onto CD, use WAV files. This insures that you will have the highest quality available if you want to re-use or re-edit the file later for a different project.

- **Post it on a host website** – Websites like MySpace.com, Garageband.com, and others allow you to store the content you've created, and allow other people to hear it as well.

MP3 or WAV?

MP3s are the most popular digital-recording format available. Why? Because they are small files, making them perfect for e-mailing, uploading to websites and general sharing. There are also some proprietary versions of MP3 files; songs downloaded from iTunes are in the M4A format, which only plays on an iPod. Microsoft has its own format called WMA (Windows Media Audio) which plays on Windows-compatible devices. But MP3 files will play on just about any brand of music player, including iPods, and computers. However, when MP3 files are burned onto a CD, the result is a data disk (not an audio CD) which may not play on all CD players.

MP3 files are small because they are *compressed*; this means that some of the data is discarded and cannot be recovered. WAV files are not compressed, which provides the best possible audio quality, but they require much more storage space on the computer and take much longer to upload/download.

Many software programs (including iTunes) convert WAV files to MP3s. This reduction in size will affect audio quality, though many people accept this tradeoff in exchange for the ability to fit more songs onto their music player. While there are programs that convert MP3s to WAV files, they do not make the resulting sound quality any better, because the data that was discarded when the MP3 file was created cannot be 'added back in'.

To summarize: Always save your initial recordings as WAV files. Use the WAV files for burning CDs, and down-convert to MP3 for distribution via the Internet.

SECTION FIVE

SUMMARY & ADDITIONAL RESOURCES

Home recording and podcasting are both constantly evolving applications. Technologies are changing quickly; people are finding new solutions to common problems; and listeners are expecting higher and higher quality recordings, even when downloaded for free.

This is why Shure continually updates the FAQ and Support sections on Shure.com. We also create and maintain our 'Shurecasts,' which are podcast versions of the most popular audio topics, from technical and troubleshooting issues to interviews with artists and engineers.

A key benefit of our Shurecasts, is that you can *hear* the difference that good techniques and specific effects will make to your recordings.

To access our extensive educational library and our Shurecasts, visit <http://shure.libsyn.com/>.

Microphone Techniques

Shure has written a comprehensive educational publication entitled Microphone Techniques for Recording in which Shure Applications Engineers describe microphone techniques and placements for a wide range of vocal and instrumental needs. A valuable resource for musicians, it is available for free online and in a printed version. See the back cover for details.



SECTION SIX

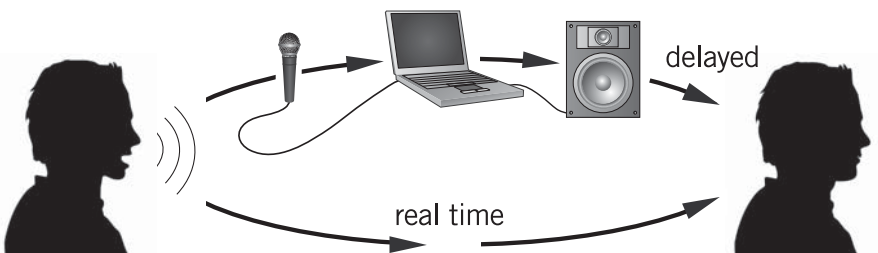
APPENDICES

Latency

Latency is a delay in the signal path caused by the time required to convert sound from analog to digital (or vice versa) or otherwise process the signal. Usually measured in milliseconds, latency can occur at multiple points in the signal path... and it can really add up. What this means is that the sound you are hearing when you listen to yourself singing might not be happening in real time. It might be five, ten, twenty, or even one hundred milliseconds *latent*.

While you cannot fully avoid latency, regardless of how much you spend for processing power, you can use hardware that allows you to listen to the sound at the beginning of the signal path.

- *If you are using one microphone...* look for a microphone with 'zero latency monitoring' (which means it has an integrated headphone amp) so you can monitor directly from the microphone.
- *If you are using multiple microphones...* use an interface box with headphone jacks, which will let you mix and monitor the combined output.
- *If you are recording directly from the sound source* (such as from an electric bass guitar connected to a direct box)... see if your USB adapter has a monitoring option.



Conversion of sound from analog to digital causes a delay (top path) called latency. The sound you hear directly (lower path) arrives milliseconds before the converted sound.

Microphone electrical outputs

The electrical characteristics of the microphone's output signal are important, because they must be compatible with the audio input to which the microphone is connected. Microphone signals have historically been *analog*, but with the increasing popularity of *digital* audio recording and editing, microphones with digital audio output have become available.

The wiring configuration of the microphone can be balanced (with two wires carrying the audio signal, with a metal shield around them connected to ground) or unbalanced (with just one signal wire and a shield). The balanced wiring configuration reduces the pickup of electrical hum or noise through the microphone cable, but the microphone, the cable, and the recorder's input must all be wired this way for this to work.

The physical connector used by an analog microphone may vary, although the three-pin XLR connector is by far the most common. (See example on page 12.) These connectors are rugged and secure, and are available with multiple pins to accommodate balanced or unbalanced wiring configurations.

Analog Output

The output from an analog microphone has three important electrical characteristics: output level, impedance, and wiring configuration. The output level or *sensitivity* is the level of the electrical signal (usually specified in millivolts or decibels) for a given input sound level. Condenser microphones typically have higher output level than dynamic mics, making them more suitable for use with recording equipment that may have inputs that are noisy or not very sensitive. Microphones with lower sensitivity require either more gain added at the mixer or interface input, or that the microphone be placed closer to the sound source. For example, dynamic microphones do not make good choir microphones because to pick up the entire ensemble evenly, they are placed at a distance that results in extremely low output. In this case, a higher output condenser microphone would be a better choice.

Most professional microphones have a low output impedance (less than 600 ohms), which allows the use of long cables (up to 1,000 feet or more) with no loss of sound quality. High-impedance microphones exhibit noticeable high frequency loss with cable lengths greater than about 20 feet. Contrary to common belief, the impedance of the microphone should not match the impedance of the input to which it is connected, but actually should be much less.

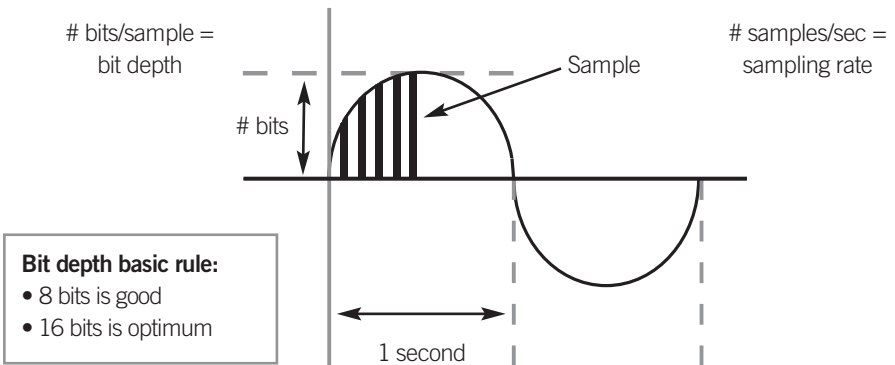
Digital Output

Microphones with direct digital output have an internal analog-to-digital (or 'A to D') converter, which changes the analog signal into a digital format. The important characteristics of this digital signal are the *sampling rate* (See diagram on page 30), *bit depth*, and *digital format*. In general, the conversion from analog to digital involves taking periodic measurements or *samples* of the audio signal level and translating those measurements into a string of 0's and 1's. You want to make sure that the software you are

using can support the sampling rate and bit depth of your digital microphone. Digital microphones offer the advantage of direct connection to a computer without the need for a mixer, sound card or external interface.

The *sampling rate* describes how many times per second the analog signal is measured. The higher the sampling rate, the higher the maximum frequency response can be. A sampling rate of 44.1 kHz (meaning that the analog signal is sampled 44,100 times per second) can accommodate audio frequencies as high as 22,050 Hertz, delivering “CD quality.” Lower sampling rates provide reduced sound quality (sometimes described as ‘speech quality’) but result in smaller file sizes and faster download speeds. Higher sampling rates are sometimes found on professional recording equipment, although there is debate as to whether sampling rates much higher than 44.1 kHz translate into audible improvements in sound quality.

The *bit depth* describes the number of digital bits used to store the measurement of the audio signal level each time it is sampled. Using more bits allows a more accurate measurement and a better quality recording, by increasing the dynamic range and reducing hiss. For example, an 8-bit sample allows the audio signal level to be measured in 256 discrete steps; if the actual signal level is somewhere between two steps, then the estimate won’t be accurate. A 16-bit sample (used on audio CD’s) allows 65,536 discrete steps, which is enough to create a very accurate estimate of the signal. Using more bits also results in larger file sizes and longer download times, however, and requires more processing power and memory when editing.



The *format* of a digital microphone’s output describes how the digital data is arranged as it moves from the microphone to the recorder or computer. The most common format for digital microphones is the USB (“Universal Serial Bus”) standard. The USB format is recognized by most computers today, and USB connectors are sufficiently compact and reliable for use in microphones. The USB connection can also supply power to the microphone, making batteries or an external power supply unnecessary. USB connections are limited to a maximum length of 5 meters (about 15 feet).

Shure Product Selection Guide

New	PG27*	Side-address cardioid condenser microphone for instrument and vocal recording applications.
New	PG27USB*	Side-address cardioid condenser microphone for instrument and vocal digital recording applications.
New	PG42*	Side-address cardioid condenser microphone for lead vocal recording applications.
New	PG42USB*	Side-address cardioid condenser microphone for lead vocal digital recording applications.
New	X2u*	USB Signal Adapter, for warm and clear reproduction in digital recording applications.
New	SM27*	Large diaphragm, side-address cardioid condenser microphone for stage and studio applications.
	SM7B	Shielded, selectable frequency response microphone delivers warm and smooth audio reproduction in close-proximity applications.
	Beta58A	Precision-engineered supercardioid dynamic microphone optimized for lead vocal applications.

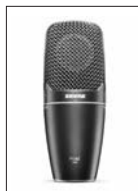
*Available 2009



PG27



PG27USB



PG42



X2u



SM27

Shure Product Selection Guide

SM58	Industry standard microphone tailored to deliver warm and clear vocal reproduction.
PG58	Professional quality microphone ideal for lead and backup vocal performance applications.
SM81	Industry-standard, flat-response microphone renown for sonic accuracy in stage and studio performance applications.
PG81	Professional quality microphone ideal for acoustic instrument applications.
Beta57A	Precision-engineered supercardioid dynamic microphone for detailed reproduction of amplified or acoustic instruments.
SM57	Industry standard, highly versatile microphone tuned for clean reproduction of amplified and acoustic instruments.
PG57	Professional quality microphone for amplified or acoustic instrument applications.

Shure Microphone Product Categories:

PG	Affordable, high quality microphones
SM	Industry standard/professional performance
Beta	Premier performance, sensitive to detail
KSM	Ultra sensitive, precise reproduction



SM58



SM81



Beta57A



SM57



PG57

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

Audio Signal Processor – See ‘Signal Processor’

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

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

Compressor – A device or software feature that controls varying signal levels by reducing the level of loud sounds.

Condenser Microphone – A microphone that generates an electrical signal when sound waves cause the spacing between two charged surfaces (the diaphragm and the backplate) to vary.

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

Delay – The time delay of an audio signal. Depending on the length of the delay and how much of the delayed signal is mixed with the undelayed audio signal, the effect can mimic the acoustic reflections that occur in rooms of various sizes, thus adding a sense of ‘space’ to a recording that suggests a particular recording environment.

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

Echo – Time delay of an audio signal that is long enough (typically more than 20 milliseconds) to be heard as a distinct repetition of the original sound.

EQ – Equalization or tone control to shape frequency response (and sound quality) in some desired way.

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

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

Frequency – The rate of repetition of a cyclic phenomenon such as a sound wave. Usually measured in Hertz (Hz).

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

Gain – Amplification of sound level or voltage.

Headworn Microphone – A microphone designed to be worn on the head.

Hertz (Hz) – A unit of measurement that represents cycles-per-second. The musical note "A" above middle "C" is equivalent to 440 Hz.

Impedance – In an electrical circuit, opposition to the flow of alternating current, measured in ohms. A high impedance microphone has an impedance of 10,000 ohms or more. A low impedance microphone has an impedance of 50 to 600 ohms.

Interface – Typically refers to a device that converts analog audio signals to a digital signal for connection to a personal computer, and vice versa. Digital audio interfaces can either be internal (on a PCI card) or external (with a USB or Firewire connection to the computer).

Latency – A delay between the time that an audio signal is converted from analog to digital, processed, and transmitted, and the time that it is heard by the listener. Latency can vary greatly depending on the software and file format used. Typical latency for audio ranges from a few milliseconds to over 100 milliseconds (1/10th of a second). If the delayed audio signal being is compared to an undelayed signal (such as a performer hearing their own voice) or to an undelayed visual reference (such as an audience member seeing the image of a live performer), delays of more than a few milliseconds can be noticeable to the listener.

Lavalier Microphone – A small microphone designed for hands-free usage. Usually clipped to the clothing.

Leakage – Pickup of an instrument by a microphone intended to pick up another instrument.

MP3 – The most popular format for compressed audio files. When an MP3 file is created, the encoding software discards some of the data that is deemed to be unnecessary or redundant. The more data that is discarded, the smaller the file size but the lower the sound quality. MP3 is an acronym for MPEG3, which is itself an abbreviation for “Motion Picture Experts Group, Layer 3”.

Multitrack Recording – A method of recording where each instrument (or group of instruments) is recorded onto a separate track and later combined into a stereo mix. Common formats include 4, 8, 16, and 24-track recording.

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

Overhead Microphone – Microphones that are typically hung from the ceiling. Common applications are choir and theater miking.

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

Pop Filter – A screen, typically made of nylon or other tightly-woven mesh, designed to prevent plosives (loud, low frequency thumps caused by the consonants “p” and “t”) from reaching the microphone.

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

Sensitivity – The electrical output that a microphone produces for a given sound pressure level.

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

Shock Mount – A suspension system for mounting a microphone that reduces pickup of unwanted low frequency sounds caused by mechanical vibration.

Shotgun Microphone – An extremely directional microphone, commonly used in broadcast and film production applications.

Signal Processor – Any device or software plug-in that can manipulate the audio signal, in terms of level, frequency, time, or phase. Examples of signal processors include equalizers, compressors, delay (echo), and reverb. Signal processors can be used to correct problems with an audio signal, or for creative effect

Sound Reinforcement – Amplification of live sound sources.

Stereo – Two channels of audio, left and right, which can be used to simulate realistic listening environments.

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

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

USB – An acronym for Universal Serial Bus, a standard designed to allow many different types of devices to connect to a computer using a standardized interface. USB also can provide power to low-consumption devices, negating the need for external power supplies. There are currently two standards: USB 1.1 and USB 2.0. For audio applications, USB 2.0 (which offers much faster data transfer rates) allows many more channels of audio to be streamed to the computer at once.

WAV – A file extension that refers to a standard for storing audio data. Commonly referred to as a WAVE file, it is short for Waveform audio format. The most common WAVE file stores full, uncompressed audio for the highest quality.

WMA – A proprietary Windows audio file format.

ABOUT THE AUTHOR

Chris Lyons is **Manager, Technical and Educational Communications**, with Shure Incorporated.

With more than 21 years of experience in training, technical support, and product management at Shure, he has presented training sessions for retail salespeople, created online training content, and supervised the deployment of third-party in-store product demonstrators, both in the U.S. and abroad. He has written numerous articles and technical papers, including *Introduction to Wireless Systems and Audio for Distance Learning*.

Additional Shure Publications Available:

Printed and electronic versions of the following guides are available free of charge. To obtain your complimentary copies, call one of the phone numbers listed below or visit www.shure.com.

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- Microphone Techniques for Live Sound Reinforcement
- Microphone Techniques for Studio Recording

Stay Connected and Keep Learning:

Shure also offers valuable e-newsletters and podcasts to help meet specific applications and troubleshooting needs. These are free of charge and can be obtained by visiting our Tech Library at <http://www.shure.com/ProAudio/TechLibrary/index.htm> or by contacting us directly.

Our Dedication to Quality Products

Shure offers a complete line of microphones and wireless microphone systems for everyone from first-time users to professionals in the music industry—for nearly every possible application.

For over eight decades, the Shure name has been synonymous with quality audio. All Shure products are designed to provide consistent, high-quality performance under the most extreme real-life operating conditions.

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