Introduction to Home Recording

Online Manual for Introduction to Home Recording
Version: 2.1 (2023-C)

Table of Contents

Introduction to Home Recording

Introduction

Recording outside of a studio - sometimes referred to as home recording or DIY (do it yourself) - is no longer a limitation to great audio. The equipment available for project studios has become more sophisticated, practical, accessible, and affordable. More and more people are getting involved with this part of the audio experience and the results are truly exciting and even groundbreaking. The best studio and most expensive gear are no longer gatekeepers getting exciting music out into the world.

Shure has been a leader in audio production for more than 90 years, and we have learned that audio quality, along with creativity and passion, and will make or break any project.

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

- Music performances
- Vocal tracks
- Overdubs
- Audio tracks for video
- ...and much more

If you are new to home recording, 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 ways to improve your results.

This 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.

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, 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.

So, What *is* Good Sound?

The term 'good sound' is subjective, but a common understanding can be thought of as the optimization of three components:

1. **Audibility** - can you hear it? More to the point, are the important parts loud enough that the listener doesn't have to strain to hear 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.

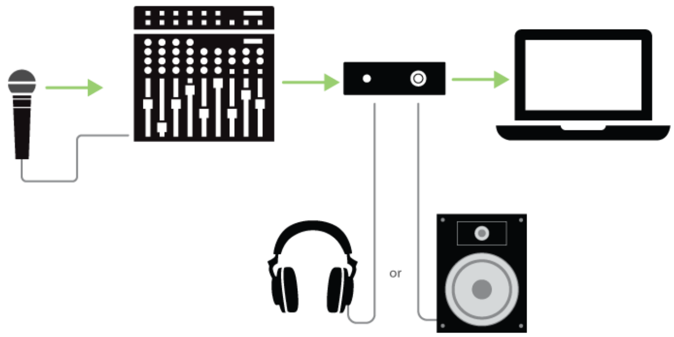
But, at the end of the day, "good sound" is mostly subjective. If you and your listeners *believe* your recordings sound good (that is, they sound like what you have in mind and people can understand them) then you *have* achieved good sound. Now comes the hard part: getting your recordings to sound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

Basic equipment needs

It can be easy to feel overwhelmed when it comes to the gear for your home recording setup. But don't worry - you can start with a simple, minimal setup and achieve great results!

Below are two examples of a basic and more complex recording signal paths for home recording.

Basic Setup



More Complex Setup

Before you create a budget and run out to buy all the popular hardware and software solutions, it's critical to match your equipment to your needs and recording environment. As you learn more about audio and what you want out of your recordings, your equipment will scale with you.

The Recording Environment

The 'deader' the better

As recording equipment becomes more compact, smaller and creative spaces can be used for your recordings. 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.

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.

Listen back with a few sample recordings

When you are recording outside of a studio, there are so many steps to take and considerations focus on to pull of a recording on your own. It can be difficult to have an educated or completely neutral observation as to how your room sounds when you are also focused on the performance or tone of your instrument. Not only that, but our brains have evolved to be quite good at ignoring the background hiss or mechanical noise of an environments so that we can focus on the important information.

How do we solve for this? Simple - make a recording of the chosen areas for a minute or so. Then listen to them on your media 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.

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.

Applications Tip: Room Treatment

We know that home recording environments do not provide the same acoustics as a professional studio. There are several ways to work with what you have and to improve it as much as possible.

The goal here is to create an awareness of the sources of potential influences on recorded sound and to provide insight into controlling them. Here are a few options to consider:

Sound Treatment

Add sound treatment to the walls and ceiling for a more even response from reflections.

Gobos for Isolation

Buy or build gobos that help to isolate instruments and microphones from other sounds in the room.

Strategic Placement

Position instruments strategically to get the best use of your space for a recording session.

Microphone Styles (Form Factors)

Microphones, which come in all shapes and sizes, are designed for different uses and settings. The microphone style, also known as the form factor, is determined by several considerations:

- **The sound source:** Will the microphone be used for a person's voice or for an instrument?
- **Source activity:**Is the sound source a person moving around or in one location? Holding the microphone or hands-free? Does the instrument involve big movements like drums or vibraphones? Or is it for subtle sounds like a flute or classical guitar?
- **Mounting:** Can the microphone be clipped directly to the instrument, held in the hand, or hung overhead?

Handheld - As the name suggests, the handheld microphone is designed to be easily held in your hand. Many handheld microphones include internal shock mounts (to guard against mechanical vibrations from movement entering the sound pickup) and internal pop-filters (to guard against plosives that occur when pronouncing some consonants). They be also be attached to a mic stand to free up your hands. 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. For this, placing the microphone on a stand is helpful.

An external shockmount is helpful when using a stand to reduce low frequency rumble from handling noise or other loud instruments in the room.

Stand mounted - Most instrument mics are designed to be attached to a microphone stand for positioning. Some are even designed in specific form factors to get closer to the instrument and out of the way of the performer. It is generally recommended to use shock mounts for instrument microphones, especially on sensitive microphones like condenser microphones.

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 a consistent distance from the microphone to the sound source.

Mobile/mountable - Some compact microphones are designed to attach directly to a mobile device like a smartphone or tablet. These mics deliver improved audio quality for audio and video recording or streaming using a wide variety of apps. In some cases, a companion app can allow mic settings to be customized.

Direct input - Some instruments, such as an electric bass guitar, are 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).

If you use a direct input, you should also consider using a monitor speaker so that you can hear exactly what you are recording.

Transducer Types

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

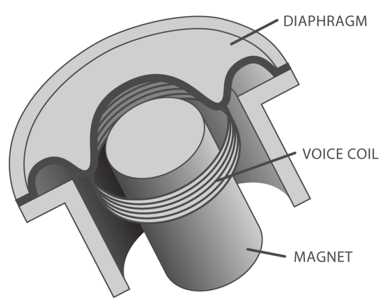
Dynamic Microphones

Dynamic mics are very rugged and can handle high sound pressure levels, like those delivered from snare drums and guitar amps, without distortion.

Dynamics Excel on Loud Sources

Loud sound sources, such as drums and electric guitar amps, won't distort a dynamic microphone. The Beta57A is shown here positioned on the snare drum.

Dynamic microphones have a diaphragm/voice coil/magnet assembly for a transducer. The assembly moves in the magnetic field, which generates an electrical signal that corresponds to the sound that is entering the microphone.



Ribbon microphone elements, a variation of the dynamic microphone operating principle, consist of a thin piece of metal, typically corrugated aluminum, suspended between two magnetic pole pieces. Ribbon microphones are highly regarded in studio recording for their "warmth" and good low frequency response. The output of ribbon microphones tends to be quite low and may require an additional pre-amplifier or booster. Note that ribbon microphones are generally not as rugged as moving-coil dynamic microphones and require more care.

Phantom power applied to the ribbon microphone could be harmful.

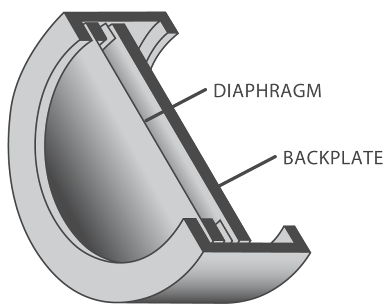
Condenser Microphones

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.

Condensers Capture Transients and Ambiance

Due to their sensitivity, condensers are often used to capture vocals, cymbals, and acoustic instruments in their most natural state.

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



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

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

If you hear distortion when using a condenser microphone close to a very loud sound source, first make sure that the mixer input itself is not being overloaded. If not, switch in the attenuator in the mic (if equipped), move the mic farther away, or use a mic that can handle a higher level. In any case, the microphone will not be damaged by excess level.

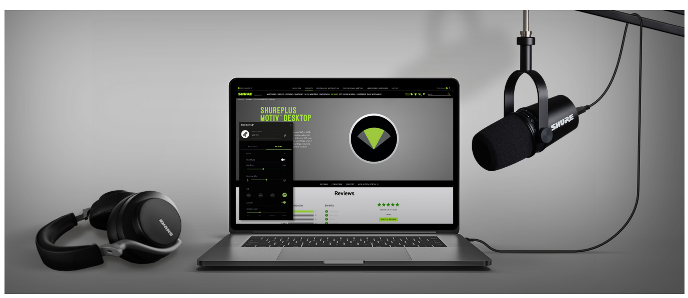
Phantom Power and Condenser Microphones

Condenser microphones require power. A common way of providing them with power is through something called phantom power. Phantom power 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.

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.

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.



Shure MV88+ Microphone and MOTIV Setup App

Some USB mics, like the Shure MV88+ provide configurable settings to get a wide range of options from a single microphnoe. The MV88+ has two microphone elements, a cardioid and bi-directional capsule to allow for stereo and mono signals, all configurable from the free MOTIV app.

In addition, some USB microphones provide a headphone output to help you monitor your recordings without latency.

To USB or not To USB

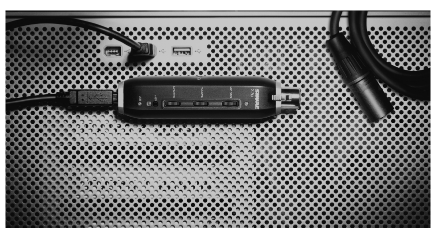
If you only need to use one microphone and plan to record directly into your computer, a USB microphone is the most convenient solution. It's quick and easy to set up, and it eliminates the need for a pre-amp, mixer, or USB audio interface.

Recording a Full Band with a Single Stereo Mic

Since group is recording live to a single stereo track, a USB microphone may be a good option.

With a single microphone for the whole room, spend extra time getting your room and instruments sound as good as possible. Then find the best spot by listening as you position the microphone in the room.

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.



XLR-to-USB Solution

The Shure X2U enables an XLR connection directly into the computer without a multi-channel interface.

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.

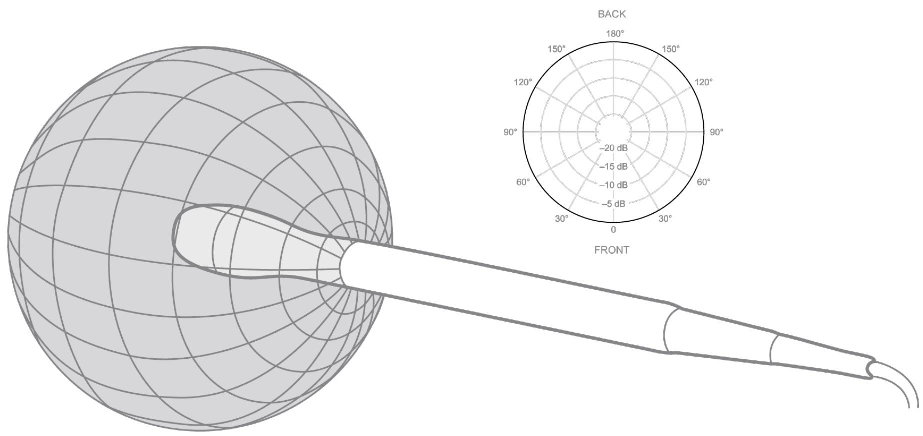
It determines how best to place the microphone relative to the sound source(s) in order to enhance pickup of desired sound and to minimize pickup of undesired sound.

The two most common directional types are omnidirectional and unidirectional.

The polar pattern of a microphone is the graphical representation of its directionality.

Omnidirectional

Omnidirectional microphones pick up sounds equally from all directions. They are good for natural room sound, group vocals, or wide sound sources like a, isolated full drum kit. Omnidirectional microphones are ideal 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.



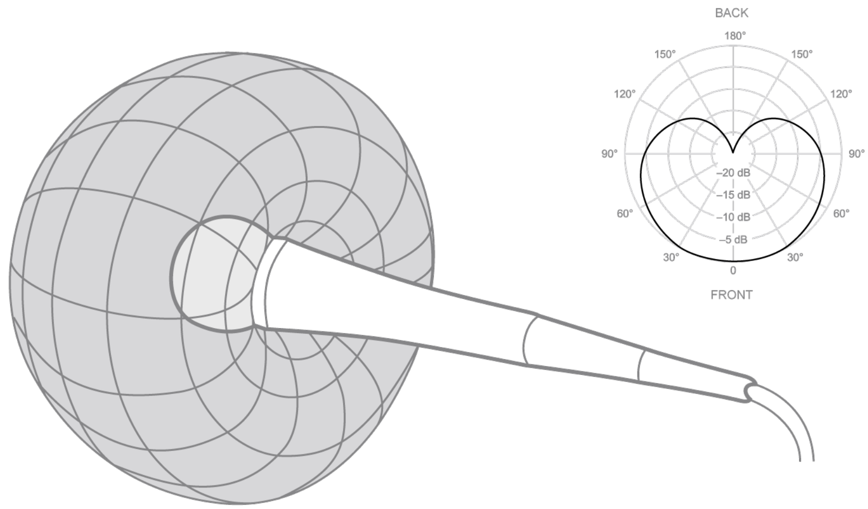
Omnidirectional Microphone

Since the microphone has the same output regardless of its orientation to the sound source, it has a polar graph that is a smooth circle as shown below. This indicates that the microphone is equally sensitive to sound coming from all directions. An omnidirectional microphone can therefore pick up sound from a wide area, but cannot be “aimed” to favor one sound over another.

Unidirectional

A unidirectional microphone, on the other hand, is most sensitive to sounds coming from only one direction, usually the front of the microphone. And, it is less sensitive to sounds coming from the sides or rear.

On a polar graph, this will appear as a rounded but non-circular figure. The most common type of unidirectional microphone is called a cardioid, because of its heart-shaped polar pattern.



Cardioid Microphone

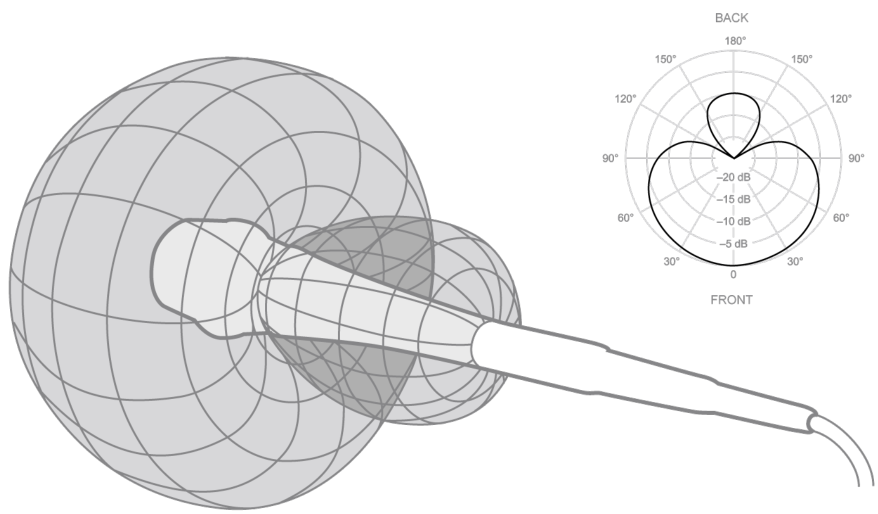
A cardioid is most sensitive to sound coming from in front of the microphone, or "on axis". It is less sensitive to sound from the sides ("off-axis"), and the direction of least sensitivity is toward the rear, called the null angle. For a cardioid pattern, this is at 180 degrees or directly behind the microphone.

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

For example, the use of a cardioid microphone for a guitar amplifier, which is in the same room as the drum set, is one way to reduce the bleed-through of drums on to the recorded guitar track. The mic is aimed toward the amplifier and away from the drums. If the undesired sound source is extremely loud (as drums often are), other isolation techniques may be necessary.

A Full Spectrum of Directionality

Unlike the cardioid, other unidirectional microphones have some pickup directly behind the microphone. This is indicated in their polar patterns by a rounded projection, called a lobe, toward the rear of the microphone. The direction of least sensitivity (null angle) for these types is about 125 degrees for the supercardioid and 110 degrees for the hypercardioid. In general, any directional pattern that has a narrower front coverage angle than a cardioid will have some rear pickup and a different null angle.



Supercardioid Microphone

The significance of these two polar patterns is their greater rejection of ambient sound in favor of on-axis sound.

The **bidirectional** microphone has full response at both 0 degrees (front) and at 180 degrees (back). It has its least response at the sides. The coverage or pickup angle is only about 90 degrees at the front (or the rear). It has the same amount of ambient pickup as the cardioid. This mic could be used for picking up two sound sources such as two vocalists facing each other. It is also used in certain stereo techniques.

CHARACTERISTIC	ONMI-DIRECTIONAL	CARDIOID	SUPER-CARDIOID	HYPER-CARDIOID	BI-DIRECTIONAL
POLAR RESPONSE PATTERN					
COVERAGE ANGLE	360°	131°	115°	105°	90°
ANGLE OF MAXIMUM REJECTION (null angle)	—	180°	126°	110°	90°
REAR REJECTION (relative to front)	0	25 dB	12 dB	6 dB	0
AMBIENT SOUND SENSITIVITY (relative to omni)	100%	33%	27%	25%	33%
DISTANCE FACTOR (relative to omni)	1	1.7	1.9	2	1.7

Comparison Chart - Polar Patterns and Directionality

There are other microphone characteristics that inherently come with directionality. Here are a few for consideration:

Ambient sound sensitivity	Since unidirectional microphones are less sensitive to off-axis sound than omnidirectional types, they pick up less overall ambient or room sound. Unidirectional mics should be used to control ambient noise pickup to get a "cleaner" recording.
Distance factor	Since directional microphones have more rejection of off-axis sound than omnidirectional types, they may be used at greater distances from a sound source and still achieve the same balance between the direct sound and background or ambient sound. An omnidirectional microphone will pick up more room (ambient) sound than a unidirectional microphone at the same distance. An omni should be placed closer to the sound source than a "uni"- about half the distance - to pick up the same balance between direct sound and room sound.
Off-axis coloration	A microphone's frequency response may not be uniform at all angles. Typically, high frequencies are most affected, which may result in an unnatural sound for off-axis instruments or room ambience.
Proximity effect	For most unidirectional types, bass response increases as the microphone is moved closer to the sound source. When miking close with unidirectional microphones (less than 1 foot), be aware of proximity effect: it may help to roll off the bass until you obtain a more natural sound. You can (1) roll off low frequencies at the mixer, (2) use a microphone designed to

	minimize proximity effect, (3) use a microphone with a bass roll-off switch, or (4) use an omnidirectional microphone (which does not exhibit proximity effect).

Understanding and choosing the frequency response and directionality of microphones are selective factors which can improve pickup of desired sound and reduce pickup of unwanted sound. This can greatly assist in achieving both natural sounding recordings and unique sounds for special applications.

Microphone Accessories

Mounting accessories can greatly help improve the ease of setup and the resulting sound of your home recordings. 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.

Recording Techniques

The following are tips to get you started on your own recordings. Remember that as your experience, equipment, and understanding of audio concepts grow, your technique will improve, along will your results. Learning to record at home is an exciting adventure, make sure to have fun and experiment as you go!

3 Tips for Any Recording

1. Record Samples and Take Notes

Try a few different recordings and see which placement sounds best to your ears. It is very hard to judge the sound that will end up on the recording during the live take. Make detailed notes as to the positions so when you listen back, you can select the best placement and position it that way every time.

2. Isolate Sounds

Use two microphones: one positioned near the singer's mouth and one positioned near the instrument's sound source.

3. Get Close (But Not Too Close)

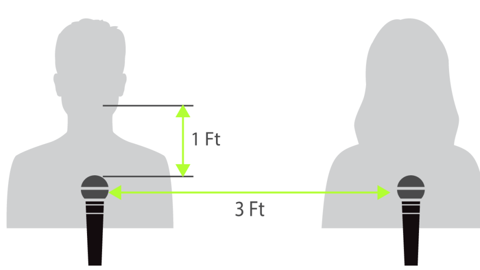
Place the microphone only as close as necessary. Too close a placement can color the sound source's tone quality (timbre), by picking up only one part of the instrument. But too far away means you will pick up more ambient sound.

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.

Checklist of Best Practices

Interested in more pointers? Here is a checklist you can use when setting up for your recordings.

1. **Microphone technique is largely a matter of personal taste.** Whatever method sounds right to you *is* right. Trust your ears and opinions and 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 not only change recording levels, but also tonal characteristics and ambient levels, all which are hard, if not impossible, to correct 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.
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.
5. **Choose the right pick-up pattern and transducer type for the application.**Microphone characteristics play a large role in how a sound source is captured. Using the polar pattern to your advantage, you can selectively avoid sounds you want to minimize, while using proximity effect and mic positioning will change the tonal quality of a source.
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.** Also referred to as isolation, the ability to edit and blend each sound independently provides more flexibility and possibilities when mixing sounds together.
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 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.

Generally, keep the microphone as close as possible to your mouth to avoid picking up unwanted room reflections and reverberation. The point where the mic sounds best is often called the sweet spot. This is usually around 6 - 12" (15 - 30 cm) from your mouth.

Do not get too close either. Proximity effect, which is an increase in low frequency response that occurs as you get closer to a directional microphone, can cause your voice to sound "muddy" or overly bassy.

Aim for the Sweet Spot

2. Aim the microphone toward your mouth from below or above.

This placement minimizes "popping" caused by plosive consonants (e.g. "p" or "t").

Positioned Slightly from Below

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 can also serve as a reference to help you maintain a consistent distance from the microphone.

External Pop Filter

4. Consider your Surroundings.

Reflections are caused by hard surfaces, such as walls, ceilings, even tabletops or music stands. They can adversely affect the sound quality captured by the microphone.

Use an Optimized Room When Possible

5. Sing directly into the microphone.

High frequencies are very directional, and if you turn your head away from the microphone, the sound captured by the microphone will get noticeably duller.

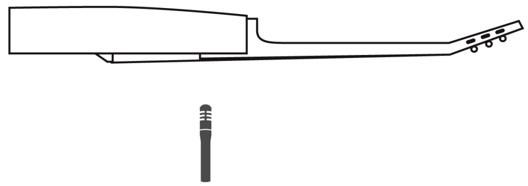
Sing Directly Into the Mic

Instrument Microphone Placement Techniques

Different instruments require different microphone placements and techniques. We can highlight some of these differences by looking at a couple of specific instruments. Again, there is no *right* solution, there are only *techniques* that provide a good starting point for your own experimentation.

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. Some people prefer clipping a miniature microphone outside of the sound hole, which allows freedom of movement.

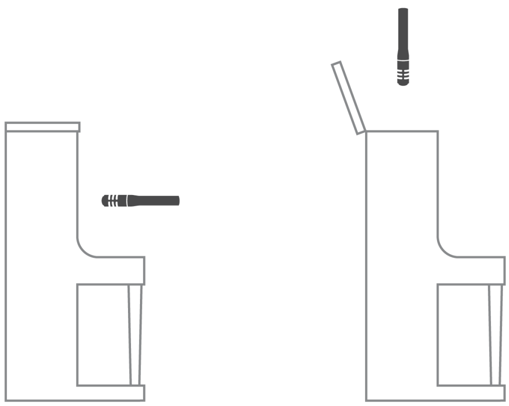


Piano

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.

For grand pianos, open the lid and aim a condenser microphone just over the top, above the treble strings.

Minimize pickup of floor vibrations by using a rubber shock-mount for the mic.



The Audio Interface and Recording Devices

The end point of the signal path is the recording device. In many home recording studios, the recording device will be the computer. 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.

Sound card	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, where you can speak into the internal microphone listen on headphones or internal speakers. But is it powerful enough for your recording needs? If not, consider a...
Audio interface	An external device that connects your audio signals with your computer, both incoming (from microphones and direct inputs) and outgoing (to headphones, loudspeakers, or other). Interfaces typically use USB cables/connectors, though there are many variations that use Thunderbolt, Firewire, Ethernet, or other connection type.
Digital recorder	A standalone, portable unit or part of a 'studio-in-a-box' solution with audio faders and effects. Most digital recording devices include simple mixing and editing. Some allow you to create both MP3 and WAV files and even convert files between the two for easier file distribution.
Analog recorder	Often associated with old-fashioned reel-to-reel systems but can still be found in the studios of musicians looking for a vintage sound. For some, simply working without a computer is an exciting and unique way of creating and working on music.

Mixers, Interfaces, or Both?

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.

A Mixer is Traditionally the Centerpiece of a Studio

However, today, especially for home recording needs, you can find far more compact mixers and, in many cases, you might be able to go 'mixerless' with an interface and recording software that allows you to make all the necessary adjustments to get the sound right.

A New Workflow Standard

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, preamps, compressors, reverbs, delays, pedals, etc.)
4. Your budget and available space

Audio 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). These days, the interface is the centerpiece of most studios, especially home studios where a mixer is not needed.

Here are a few ways that using an external audio interface helps your home studio:

- **Audio Inputs:** Interfaces typically have 2 or more audio inputs. The number of inputs equals the number of live, individual tracks that can be recorded at once. For example, if you have a 4-input interface, you can dedicate all four inputs to individual drum microphones. Or you can record a guitar on track one, bass on track two, and a stereo mix of your drums on tracks three and four.
- **Audio Outputs:** To listen back to your recording takes and work on the post-production, listening is critical. Interfaces usually have dedicated outputs that are connected to your monitor speakers. Sometimes these outputs are also used to re-amp sounds (send a DI guitar to effects pedals and an amplifier) or to use outboard effects for a more professional or vintage sound.
- **Headphone Output:** Dedicated headphone jacks allow you to monitor input sounds without latency (delay from the computer) and adjust the levels quickly without using the software.
- Processing Power: The interface itself has processing power on-board that allows you to do more audio than a typical computer soundcard.

Why Would I Need a Mixer?

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.* This may be useful if your interface only accepts one or two inputs at a time. Let's look at a few common examples:

Multiple Drum Mics	To get a detailed drum sound, multiple microphones can be set up all over the kit to capture each drum in close proximity. These microphones are all connected to the mixer, where the overall sound is mixed together. A mono or stereo output can be sent to the interface with minimal inputs.
Backup Singers	While the main vocalist is recording the track, multiple background singers can be blended to a single track, which is then routed to the interface.
Live Effects	Using a mixer can open up possibilities to incorporate live effects with outboard equipment. Simply connect the microphone to the input and use the aux sends to blend in reverb, echo, or other effect before sending the final signal to the interface.

For these and other creative use cases, you will want to consider getting an external mixer (or 'mixing console' or 'mixing board'). Mixers come in many varieties, sized, and features. It can get expensive and technical very quickly - so make sure you get something that is useful but also fits comfortably into your home studio.

Most importantly: be sure that the mixer you choose has enough inputs to handle the number of microphones you are using. Also 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.

If you wish to capture the sound of each microphone on a separate track (providing options in post-production), then your mixer will need to have direct outputs for each channel to connect to your recording interface or device. Otherwise, you will only be able to capture the blend (mix) of all of the inputs.

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 or coloration. Some microphones, like ribbon microphones and some dynamics, require more gain boost to get to the right input level at your interface. While pre-amps exist in mixers and interfaces, standalone pre-amps are common for both studio and home recording to achieve a better sound. It often does a better job of rejecting electrical noise and hum than the internal one in the audio interface and some mixers. A mic preamp usually accommodates just one or two microphones and offers separate controls and outputs for each.

When your audio interface just can't amplify the signal enough without adding excessive hiss, consider using an external pre-amp to cleanly increase the signal strength. Additionally, it is common that recording interfaces have extra line-level inputs in addition to the one to two XLR style inputs that work with microphones. An external pre-amp lets you take advantage of these additional inputs.

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. Recording software is all about mixing your sounds together, effects processing, and exporting your final mix to be mastered or shared online. 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.

With so many software options these days, it's most important to choose one that you feel comfortable with and enhances your workflow. Some software is known for creating beats or loops, while others are known for editing, mixing and mastering.

Make a list of what you are planning on doing and make sure the software you choose has all that functionality.

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. 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 consoles as plug-ins.

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.
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 or theater. Reverb occurs naturally but 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. For example, you might want the singer to sound like she is in a subway station or distant hallway. Adding the right amount of reverb will allow you to achieve this effect.
Delay or Echo	Delay and echo works by adding intentional delay and repeats 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 or can be used to create new textures and space in your recording.
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.

Playback and Listening

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 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**, it is best practice 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 add unwanted sounds on your new track, and even cause feedback in some cases.

The Importance of Real-Time Monitoring

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. 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.

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 the stereo system they use to listen to music, 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 as accurate.

Studio Monitors for Listening and Post Production

Headphones or earphones - Headphones are another indispensable tool for home recordists. Headphones allow you to listen closely to your audio tracks and monitor realtime signals in addition to pre-recorded material (also known as overdubbing). They are easy too take on and off so you can quickly start or stop a take, talk with your collaborators, or walk away from your computer without changing any settings. 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 or risk your recorded material getting picked up by the microphone.

Headphones for Overdubbing

Selection Guide

Shure Microphone Product Categories

Before we get into specifics of popular microphones, here is an overview of the product categories Shure offers.

MOTIV™	Digital microphones and recording solutions
PGA	Affordable, high quality microphones
SM	Industry standard/professional performance

Beta	Premier performance, sensitive to detail
KSM	Ultra sensitive, precise reproduction

Shure Product Selection Guide

XLR Microphones

Vocal Microphones

PGA58	Professional quality microphone ideal for lead and backup vocal performance applications.	
SM58	Industry standard microphone tailored to deliver warm and clear vocal reproduction.	

BETA 58A	Precision-engineered supercardioid dynamic microphone optimized for lead vocal applications.	
SM7B	Shielded, selectable frequency response microphone delivers warm and smooth audio reproduction in close-proximity applications.	

Instrument Microphones - End Address

PGA57	Professional quality microphone for amplified or acoustic instrument applications.`	

SM57	Industry standard, highly versatile microphone tuned for clean reproduction of amplified and acoustic instruments.	
BETA 57A	Precision-engineered supercardioid dynamic microphone for detailed reproduction of amplified or acoustic instruments.	

PGA81	Professional quality microphone ideal for acoustic instrument applications.	
SM81	Industry-standard, flat-response microphone renown for sonic accuracy in stage and studio performance applications.	

Instrument Microphones - Side Address

PGA27	Large diaphragms cardioid condenser microphone for instrument and vocal recording applications.	
SM27	Large diaphragm cardioid condenser microphone for stage and studio applica-tions.	
PGA181	Professional quality instrument microphone, offering an unobtrusive presence as a side-address microphone.	

BETA 181	precision-engineered, versatile small-diaphragm condenser instrument microphone with interchangeable polar pattern capsule options.	

USB Microphones and Interfaces

MV7	Digital dynamic microphone with both USB and XLR outputs for use with computers and professional interfaces alike. Connect via USB and explore additional set-up features and Auto Level Mode within our ShurePlus™ MOTIV app. Includes headphone output for realtime monitoring.
MV88+	Stereo condenser microphone that enables crystal clear, mobile recordings in adjustable stereo widths, including mid-side or cardioid options. Includes headphone output for realtime monitoring.
MV51	Digital large-diaphragm condenser microphone for home studio recording and podcasting.
MV5	Digital condenser microphone with integrated stand for instant setup. Record directly into your computer, laptop or mobile device.
MVi	Digital audio interface connects a professional XLR microphone or 1/4" instrument output to a computer or mobile device. Features include 5 DSP preset modes for EQ, compression, limiter settings, gain control, mute, and volume, and phantom power.

X2U	XLR-to-USB signal adapter connects any XLR microphone to a computer for recording with headphone monitoring. Features include USB connectivity, integrated preamplifier with Microphone Gain Control, and Zero Latency Monitoring for real-time playback.

Headphones

SRH240A	Perfect for general listening, offering excellent sound reproduction and comfort.	
SRH440A	Optimized for monitoring and accurate listening, offering professional sound quality and comfort.	
SRH840A	Premium headphones optimized for studio recording and critical listening.	

Shure Microphone Selection Guide

As discussed previously in this guide, you will get the best results by experimenting with placement and listening critically. Microphone selection is another important consideration for getting the right sound. Below are some suggestions of Shure microphones for certain instruments and sound sources. As always, we encourage you to get creative and listen closely; over time you will better understand each microphone's characteristics and your preferences in your recordings.

Vocals

SOURCE	**IDEAL MIC**	**ALSO TRY**	
Solo Vocal	SM7B	SM58	PGA27

Ensemble/Choir	MV88+	PGA27	PGA181
Spoken Word (Broadcast/Podcasting/Streaming)	SM7B	SM58	PGA27

Instruments

SOURCE	**IDEAL MIC**	**ALSO TRY**	
Electric Guitar (Amplifier)	SM57	PGA57	PGA181
Acoustic Guitar	PGA181	SM57	PGA81
Electric Bass (Amplifier)	PGA52	PGA181	SM57
Acoustic Bass	PGA52	PGA81	SM57
Piano	PGA81	PGA181	SM57
Strings	PGA27	MV88+	PGA81
Woodwinds	SM27	PGA81	MV88+
Brass/Saxophone	PGA27	SM57	PGA81
Harmonica	520DX	SM58	PGA58

Drums

SOURCE	**IDEAL MIC**	**ALSO TRY**	
Kick Drum	PGA52	SM7B	PGA57
Snare (Top)	SM57	PGA57	PGA181
Snare (Bottom)	PGA81	SM7B	PGA57
Rack/Floor Toms	PGA56	SM57	SM7B
Overheads	SM81	PGA181	PGA181
Auxilary Percussion	PGA56	PGA57	MV88+

Stereo Techniques

SOURCE	**IDEAL MIC**	**ALSO TRY**	
X-Y	SM81	PGA81	PGA181
M-S	MV88+	BETA 181 (pair)	
Spaced Pair	SM81	PGA27	PGA81

Saving and Sharing

Saving and 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 saving a backup (or two) is the only way to ensure your files are safe. Home recordists should have a number of backup solutions for storing their recordings:

- **Backup to an external hard drive or flash drive** - Music projects get large very quickly and therefore harddrives and external storage are essential for protecting your recording files, including the numerous takes and individual tracks you use to build your final mix. In this case, the biggest (in GB) you can afford, the better.
- **Post it on a host website** - There are numerous websites that allow you to store the content you've created, and allow other people to hear it as well. Beyond simple cloud storage options, sites like Bandcamp and Soundcloud allow you to post and manage your material directly, enabling a direct line of access to your audience and community. Meanwhile, there are numerous professional streaming services that leverage mass distribution and allow people from all over to easily discover and access your material.

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 iOS device. 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 iOS devices, 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'.

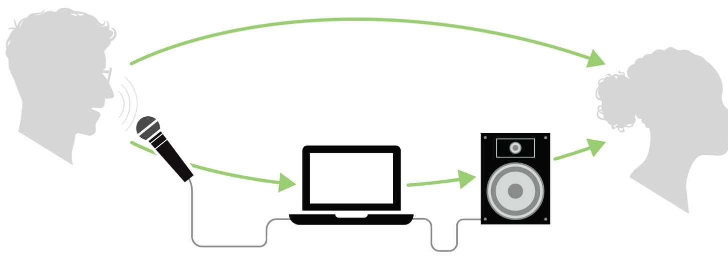
To summarize: Always save your initial recordings as WAV files. Use the WAV files for burning CDs, and down-convert to MP3 for distribution via the Internet.

Learn More

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. It can really add up to a noticeable degree, depending on your signal path, processing blocks, computer processing power, and more.

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 behind due to the long route from voice to headphones. The higher the latency, the more noticeable and problematic the outcome.



Latency

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

While you can never fully avoid latency, 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 'realtime headphone 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.

Microphone Output

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.

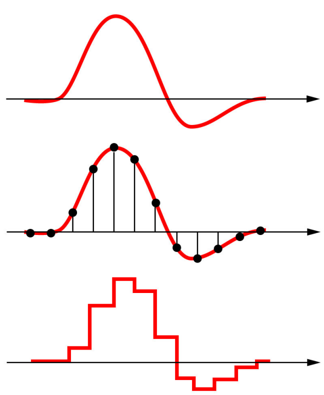
Most professional microphones have a low output impedance (less than 600 ohms), which allows the use of long cables (up to 1,000 feet or more) with no loss of sound quality. High-impedance microphones exhibit noticeable high frequency loss with cable lengths greater than about 20 feet. Contrary to common 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).

Reference Information

Glossary

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

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 very 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 com- pared 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 micro- phone 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.