# Recording Your Voice at Home

Most homes aren't terribly well-designed for recording, but with a few household items and a small amount of effort, you can produce some excellent recordings of your voice to create voice-overs for narration.

### Don't I Need a Recording Booth?

Nope, and who'd want that sitting around the house during the rest of the day, when you're NOT using it? The truth is, most recordings made at home will sound hollow, distant, "boxy", and otherwise not that great, unless you take some basic steps to improve your recording environment. But you really can get good results without spending a fortune.

Most homes are made for living, not recording, and so they have attributes that aren't helpful for recording...think rooms with 90-degree corners and parallel walls...these cause problems known as "standing waves", which will let your voice bounce back and forth between those walls over and over, ensuring that your recording will not just capture your voice, directly out of your mouth, but also those sound waves that leave your mouth and bounce between the walls. This all happens so quickly you won't notice it (there's no repeating echo like you're above the Grand Canyon...remember, this is your living room or bedroom!), but when these waves reflected off the walls are blended into your recording alongside the waves coming directly out of your mouth, it partially cancels them out (a process known as "phase cancellation", leaving the sound of your voice thin and reedy upon playback.

### Oh, No! But You Said I'd Be Able to Get Decent Recordings at Home, So...How?

The way you get around this is to keep the sound from bouncing off any flat, reflective surface near you while you're recording. Absorption is the method that most recording studios will use to prevent any reflected sound from being part of the recording, ensuring that only the original, intended signal is recorded. So if you're set up at the dining room table, be sure to put a towel or something on the surface (to prevent reflections from the tabletop itself), and look for a comforter or duvet...something that's got some padding and thickness to it. Hanging these on the walls to your left and right sides will keep you from getting reflections from those directions. Depending on your distance from the wall you're facing, you may need one there as well (if you're a good distance away, you might not, but do some test recordings with and without to see which gives you better results.) Finally, most of us forget, but reflections will happen in any direction, and any room you're in at home is usually some sort of cube, so...remember that your ceiling can and may work against you here as well. (Textured ceilings like popcorn, etc., don't really do a significant amount to break up the sound waves, so you'll still get some reflection from them unless they're fairly high.)

### Okay...I've Put Together Some Amateur Acoustic Treatment. Now What?

Congratulations, you've now made an environment to improve your recordings. Now you'll only need to think about the rest of this process...how to get your voice into the computer, so that you can do what you want with it. For this, you'll need at least a USB microphone, or else an audio interface, microphone and cable, to get your voice recorded and converted into a digital file. USB microphones come with a cable that plugs into a regular USB slot on your computer, where an audio interface is like an external sound card for your computer, but one with much better components and which provides connections to pro audio gear (XLR microphone cables and ¼ instrument cables, vs. 3.5mm microphone and headphone inputs on a cheap, standard computer sound card.) The analog-to-digital converters (ADCs) and digital-to-analog converters (DACs) on an audio interface (even a fairly basic, $100 model) will be many times better than those in your computer's sound card, and if you listen to .mp3s or other compressed audio formats, you'll notice a real improvement when those files are played back through your audio interface as well.) Most professional voice work is done using condenser microphones, as they provide a wider frequency range and greater sensitivity, but certain dynamic microphones can provide very good results as well. Condenser microphones do require 48-volt phantom power for the microphones to function, so if you decide to use something other than a USB microphone or a mic/audio interface solution, be sure you're adequately prepared.

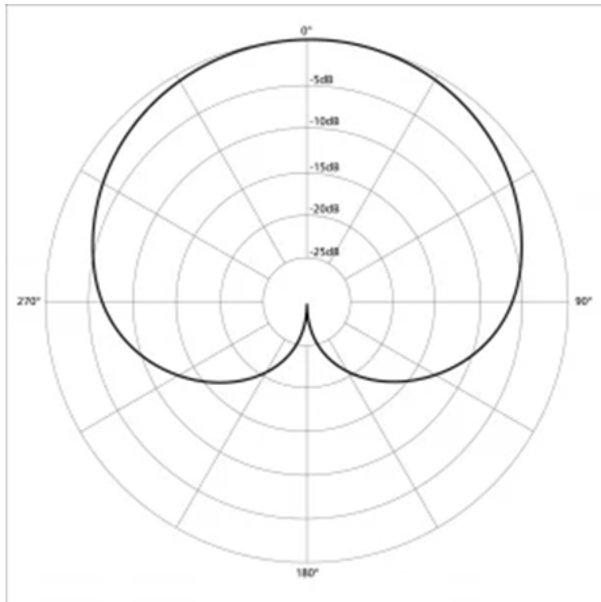
**Consider a Pop Filter**

A pop filter can also be tremendously helpful here, to keep the plosives in your speech from overloading the microphone capsule (think of a speaker who said a word that starts with the letter "P", and if the bass overload practically sends everyone running, or sounds as thumpy as a bass drum,  you've got the idea.) Pop filters can be purchased fairly inexpensively ($15-25), but you can also create your own with a wire hanger and a pair of panty hose, and when placed between your mouth and the microphone, this will do a very credible job of stopping those pops and plosives from becoming part of your recording.

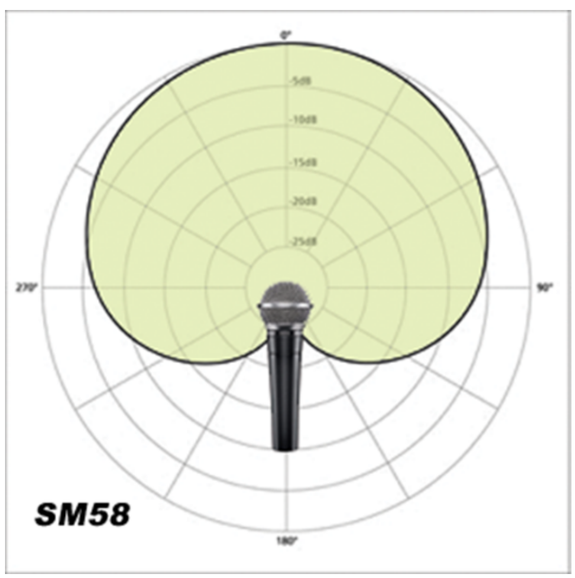
It IS possible to record without the use of a pop filter, but you need to be smart about it...positioning your mouth so that your speech goes right down the barrel of the mic (or right into the capsule of a side-address condenser microphone) practically guarantees that every expulsion of air, voice and (yes) even saliva will go into your microphone...probably not what you had in mind, right?

**Learn and Use the Polar (Pickup) Pattern of Your Microphone**

Microphones all have what's called a polar pattern, and you can think of this pattern as "where the microphone is sensitive, and hears best." There are several kinds of polar patterns, but for voice work, the most useful is cardioid, meaning "heart shaped" (look at it upside down in this image):




This means that, on a dynamic microphone like the SM58, there's a small zone immediately around the ball (which is itself a pop filter, but not a great one) where the microphone's sensitivity is the highest:

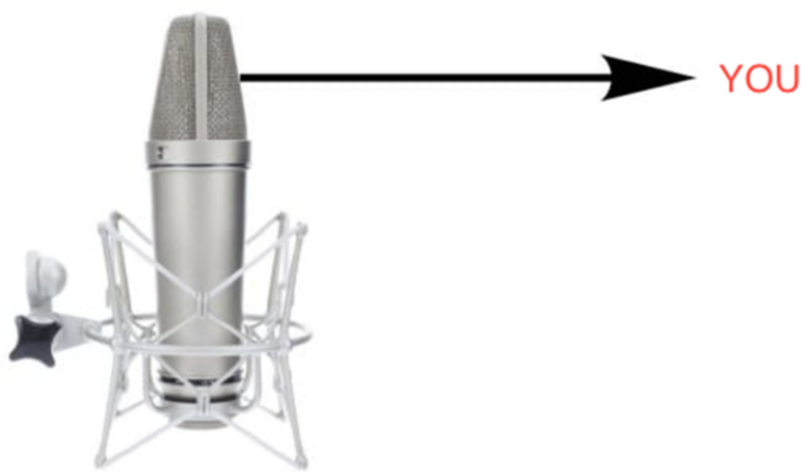



You'll notice that the microphone's sensitivity is the lowest at the base of the microphone, where its cable plugs in. This means that sounds coming from "behind" the microphone are picked up the least, and this is where rejection is the highest.

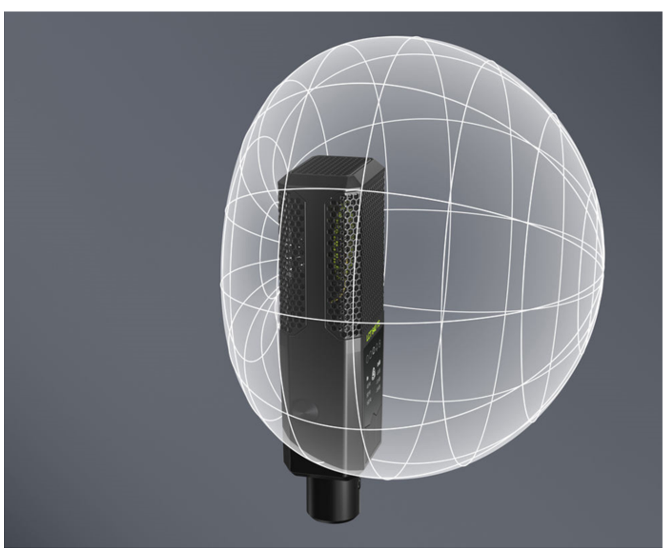
But polar patterns are sometimes easier to grasp with dynamic microphones when you're new to them. Nearly all condenser microphones also utilize a cardioid polar pattern (many also include anywhere from 1-8 other patterns, changeable via a switch on the microphone itself!), but as you don't "point" these microphones in quite  the same way you do with dynamic microphones like the SM58 above, it's not always obvious where the polar patterns fall on condenser mics.

**Come in Through the Side!**

Most condenser microphones, due to the way the mic capsule is positioned inside the body of the microphone, are something known as *side-address* mics. Whether cylindrical or flattened on opposite sides, these microphones may look like a rocket ship, but they don't pick up sound from what looks like it's up *above* the microphone...they pick up what's on one *side* (some microphones can even pick up from both sides, but using them isn't all that common in voice work.)




As a result, that same cardioid polar pattern still applies, but it applies from the *side* of the microphone:



**Where Should I Be?**

How far away from the microphone you should be when you record is up to you, and the sound you're looking for, but remember that being too far away may result in your recording *sounding* too far away. This puts more of the room into your recording as well, vs. when you're up close on the mic. Also, most microphones exhibit a characteristic known as proximity effect, meaning that the closer your mouth to the microphone, the more bass response the microphone will add to your voice. (FM DJs leaned pretty heavily on this to sound cooler than everyone else in the room back in the 1970s, and it's fun to try, to help you find what you think of as your own voice's sweet spot. Try to find something that you like and want to repeat, and then note where you are in relation to the mic, pop filter (if applicable), etc., so that you can pick right back up where you left off last time, still in your favored position.

**What Software Do I Need?**

Any sort of Digital Audio Workstation (DAW) software will do, but be aware that these can run anywhere from free to $700-ish or more. If you've never used a DAW before, and you don't see yourself ever needing to do more than the occasional voice-over, you're probably much more on the free end of this spectrum. If you're hoping to do more advance voice work, and/or you are a musician, you'll probably want to get into one of the more professional DAWs to get more features and plug-ins suitable for higher-end production.

**Some good free DAWs/audio editors are:**

PreSonus Studio One Prime (PC and Mac)

Audacity (PC and Mac)

Garageband (free, Mac only)

Tracktion Waveform Free (PC and Mac)

Avid Pro Tools First (PC and Mac)

Cubase LE (PC and Mac)

**Some good paid DAWs are:**

Adobe Audition (PC and Mac, available by monthly/annual subscription only, approximately $21/month)

Reaper (PC and Mac) - $60 for 1-seat personal license good through next full version upgrade

PreSonus Studio One Professional (PC and Mac, available as both perpetual license and monthly/annual subscription)

Avid Pro Tools Artist (PC and Mac, available by both monthly/annual subscription)

Avid Pro Tools Studio (PC and Mac, available by perpetual license or monthly/annual subscription)

...and there are plenty of others as well.

**OK, I Have A DAW – Now What?**

Once you've installed your USB microphone (most are class-compliant devices, meaning they don't need a driver to use them, but should begin to work shortly after being connected to the computer) or audio interface (most do require drivers, at least on Windows computers), you'll need to open up your chosen recording software, and let it know about your audio hardware setup. Look for a menu option called Settings or Preferences, and set your microphone/interface as the audio recording device, and for audio interfaces, set the interface as the audio playback device, and save your updates. Now your software and hardware should be able to work together.

**What Recording Specs Should I Use?**

At High Plains Public Radio, we use the same specs used for audio CDs:

- 44.1 kHz sample rate (also sometimes shown as 44100)

- 16-bit bit depth

These are the biggest settings that affect the quality of your recording.

**Sample rate** refers to the number of audio samples recorded every single second that you're recording (so in this case, it means 44,100 samples are recorded each second)...we refer to "cycles per second" as Hertz. This also means a sample rate of 44,100 (or displayed as 44100) can be thought of as 44.1 kiloHertz (or kHz). Sample rate directly affects frequency response, and along with bit depth and channel count, it has a direct effect on the resulting file size.

**Bit depth** (also known as word length) is a measure of the number of bits in each sample, and corresponds directly to the resolution of each sample, and also affects the ability of the file to capture *dynamic range,* a term that refers to the difference between the quietest and loudest parts of a signal. A bit depth of 16-bit integer resolution provides for a dynamic range of 96 decibels (abbreviated as dB), which is quiet sufficient for capturing your voice recording.

**Channel Count/Number of Channels** will usually refer to mono or stereo (though many DAWs can also work with multichannel audio, such as 5.1 or 7.1, or some of the newer Dolby Atmos configurations up to 24.1.10.) Remember, since you only have one mouth, which will be recorded via a single microphone, the most sensible choice from a space-saving perspective is mono. So while it doesn't mean the end of the world if you record your voice in stereo, it WILL result in a file size that's exactly double what it really needs to be, and for literally no additional benefit. Keeping this in mind when recording can help prevent using additional storage, both during production of your project, and also archiving the final product later.

**OK, What ABOUT Mono vs. Stereo? How does that affect me?**

Mono and stereo audio are both options for playback at HPPR. But remember there's a different between the files you're recording and using to produce your project, and the specs of the final resulting file. As mentioned above, it only makes sense to *record* your voice in mono, for the purpose of using less hard drive space for something that can be heard in both ears anyway, and which will not have any different program material on the left side vs. the right side.

BUT, if your project involves blending your voice with other audio files, such as those containing music (maybe the theme song for your feature, or even just a plain old music bed under your voice), that music is probably in stereo, and so rendering your final result into a mono file won't sound as good...a good rule of thumb is to use stereo for that final project file if any of the elements in your production started off as stereo files, and leave the final project file in mono if not.

**How Do I Get the Hang of the DAW?**

Honestly, the best way to learn it just to use it.

Spend a little time recording yourself, and use the Help files for the software if you run into any difficulties. The big red circle button is generally the one that will begin the recording, the button with the black square is the stop button, and the green or black triangle button is usually for playback (the space bar will stop and start audio playback in nearly every DAW, for what it's worth). If your voice is sending the input meter up into the yellow and red sections, you can probably turn down the input gain, to minimize the chance of clipping from an input that's too “hot” (loud). If you can barely see your meter moving at the bottom (or left), however, that may be an indication that you need to turn the input gain UP, to bring a stronger level into the computer for recording.

Once you’ve done a few recordings and played them back, learn a bit about how to navigate forward or backward in the timeline of your recording, and spend some time figuring out how to edit the audio waveform that you see...all DAWs have this capability, and it’s usually fairly simple to figure out (Help files can also be of assistance here as well.) Make a point of learning how to remove portions from the beginning, end and middle of your recording, move segments forward or backward, and also how to apply a fade in at the beginning of an audio clip, and a fade out at the end. You’ll begin to see how easy it is to assemble good-sounding audio, without extra noises around your edits that have been caused BY your edits. And these skills will become the basic tools by which you will craft your final audio product.

**Anything Else I Should Know?**

Oh, yeah, definitely. Here are a few things:

• **At HPPR**, our audio is played back by an automated audio server, directly over the air. This means we have to ensure that the audio we play back always meets certain standards, both for over-the-air quality, and for timing purposes. So in addition to ensuring that your project meets the 44.1 kHz, 16-bit, mono/stereo requirements noted above, we will also ask you to:

- Leave one second of silence at the beginning of the final rendering of all projects you produce;

- Edit the record to .3 seconds less than its specified air length (i.e., a 4 minute segment would actually be 3 minutes, 59.7 seconds when completed);

- See previous bullet point, and leave .7 seconds of silence at the end of the final rendering of your file.

• **It’s not uncommon** for your final project, after completion, to sound quieter than other audio files on your computer, such as podcasts, ripped or downloaded songs, etc. After all, being sure not to clip your inputs with too strong a signal makes it easy to lean a bit too far in the opposite direction, recording a very quiet signal, thus boosting the sound of the room versus your voice, and resulting in a quieter final product overall. Be aware that your final production can still have its volume raised before we broadcast it, so that it won’t sound quieter than the other items aired before it and after it.

• **Applying effects** such as compression, equalization, limiting or other changes is certainly possible (depending on your DAW; nearly all offer these features, but only some allow you to hear the results of your tweaking in real time.) However, unless you’ve had some training or experience with these tools, it’s best to leave them to someone further downriver from your part of the production process. If you DO decide to experiment with these, that’s just fine, but please be sure to make a copy of your final feature and experiment on that, to help ensure that you don't negatively affect the file you plan to send in for broadcast.

• **Backing up your project** and saving your audio files and projects frequently, is never a bad idea, and most of us have access to at least a small amount of cloud storage. If you believe that changes may need to be made in your project, or simply want to back it up somewhere other than the computer where you produced it, consider uploading these files and other project components to Google Drive, Dropbox or another cloud storage option. This step is frequently the difference between reopening the project to make a small, quick change, or having to redo the project entirely if changes are needed. Some producers will even save new versions of a project with the date in the filename, in order to be able to go back to an older version for any reason. If you incorporate the date in YYYY-MM-DD format, at the same point in the filename, all of your files can be sorted by name and still be sorted by date as well. I also like to add some text at the end that tells me what I changed in that version...that also helps if you decide to retrace your steps and go back to an earlier version, to create a new branch and direction for your project.