Standardize Your Voice Over Home Set Up

A Basic Guide for Voice Over Artists

Collaboration Effort By:

Geoff Bisente
Eddie Correa Jr.
Christian Banas
Benjamin Isaac Diskin

Preface / Introduction

COVID-19 has swept up the world as of late. Studios are finding themselves having to close their door and actors are social-distancing as best they can. The industry has come to somewhat of a standstill, but at the same time, we still have deadlines to meet for clients. In the meantime studios are looking into "work from home" solutions in order to keep the industry moving forward. Auditions not only include a performance audition, but a "Broadcast Quality" home audition as well. Some clients have even gone as far as to urge Not to audition if you do not have a home setup.

- Talk about Scarlett and Expectations
    - Address shit now. Shut up. Sit down.
- This is not like the end all be all, but rather, this is our way of trying to close the gap between home set up qualities.
- You don't have to listen if you don't want to.

The Equipment

Part 1: Room Acoustics

Every professional in the studio world will tell you that a well-treated room will make any microphone sound great. It doesn't matter if you have a Neumann TLM 103 or a Blue Yeti, they will sound bad if your room is bad. That is why it is crucial to focus on treating your recording space first before buying any type of interface and/or microphone.

In a perfect world, every single recording artist would have an airtight room filled with panels for absorbing, panels for diffusing right in their own homes. However, that is not practical and we understand this. Our goal is not to help you create a professional studio, but instead, to create a "dead" space. What we mean by a "dead" space is "to create an environment where *no sound is able to reflect and bounce around*." The reason a dead space is crucial is that it will prevent echoes from happening. Echoes are a mixer's worst nightmare and are near-impossible to edit out in post. Now, how does one create a "dead" space to record in?

Provided below are a few resources you can research to help treat the space you are recording in to have that "dead" space we are looking for.

PRODUCT	CATEGORY	DESCRIPTION
Studiobricks	Booth	Mobile, easy to put together and take apart. No screws or tools needed. State-of-the-art acoustic treatment. Flexibility with additions to the booth. Double-Walled and Triple-Walled options. Project Studio Tier to High Commercial Grade Home Studio.
Vocal Booth TM	Booth	Low commercial tier home studio option. Good, solid booths with decent acoustic treatment and add ons. Long-established company that started the standard of home booth recording.
Audimute Sound Absorption Blankets	Acoustic Treatment	Really good sound absorption blankets. Perfect for making a PVC booth out of or just lining your walls.

2+ Inch Pyramid Foam	Acoustic Treatment	Pyramid foam because it is better at sound diffusion. 2+ inches as the more you have, the more sound you absorb.
Ultra Thic(c)k Moving Blankets	Acoustic Treatment	The thicker the blanket, the better. They are good if you need to make due with what you have.
Vocal Booth To Go	Booth	A good starter point. A high-grade blanket fort. Great with the kids.

Again, we cannot stress how important it is to prioritize your room acoustics. The cleaner the sound, the easier it is to mix. Think of your sound like a painting canvas, which one would you rather have: a blank canvas or a canvas with random paint marks everywhere? That's how it is for audio engineers. If you cannot hear the differences in what a treated room does for your sound, we have pre-recorded some samples of what they may sound like at different price points provided here.

Part 2: Audio Interfaces/Preamps

After testing to make sure your recording space has cleaner acoustics, the next step would be to address your Audio Interface. Now, there is some confusion between the terms "audio interface," "preamp" and "channel strip." We want to make sure we can clear that up for you before moving forward.

Audio Interface

- A device that directly plugs into your computer that takes the analog recording of your voice, converts it into digital data and sends it to the Digital Audio Workstation (DAW) that you are using.
- The Audio Interface acts as an "easy all-in-one" component that does multiple jobs in one device.
- Jack-of-all-trades, master of none.
- A typical audio interface analog flow looks like:
    - Talent -> Microphone -> Audio Interface (serves as Preamp, Processor, A-D Converter) -> DAW -> Studio Monitors/Headphones

Preamp

- A device that takes the audio and amplifies it. That's pretty much it.
- An audio interface can do a preamp's job, but a preamp cannot do all of the audio interface's jobs.
- A typical preamp analog flow (aka what you need in order to make it work) looks like:
    - Talent -> Microphone -> Preamp -> Processor -> A-D Converter -> DAW -> D-A Converter -> Studio Monitors

Channel Strip

- A single channel in a mixing board.
- Think of a channel strip as like the middle-ground between a preamp and an audio interface. It is not necessarily the one-focus preamp, but it is not the all-in-one audio interface. **A channel strip amplifies the sound and is able to provide equalizer (EQ), compression and other items, depending on the mixing board capabilities.**
- Most companies have taken the channel strip concept and have made standalone channel strips, due to popular demand.
- A typical channel strip (Standalone) analog flow looks like:
    - Talent -> Microphone -> Channel Strip -> A-D Converter -> DAW -> D-A Converter -> Studio Monitors

As the talent, you don't necessarily need anything other than an Audio Interface, so we will focus on that for this document.

As stated earlier, audio interfaces take the analog voice data and converts it to digital data. It is during this process that an audio interface can have the highest influence on the quality of your audio. Cheaper audio interfaces tend to heavily color the analog data, sometimes leading to more artificial sounding recordings. The higher in price point you go, the better the conversion from analog to digital.

Instead of having everyone buy interfaces, try them out and return them before Amazon's 30-day return policy, we have provided a list of audio interfaces we recommend at various price points! They have been provided below:

PRODUCT	MODELS	PRICE	ANALYSIS
Universal Audio Apollo Twin	Thunderbolt (Mac) USB (Windows)	$900	A great interface that has a lot of versatility. It typically comes with a package of plugins that can emulate the tonal properties of other pres. If you are looking for versatility and like to play with your settings then this is perfect for you. Digital recreations of other popular pres are also available for purchase and are a fraction of the hardware originals.
Universal Audio Arrow	Thunderbolt 3 (Mac/USB C)	$500	This is a smaller, more affordable unit from Universal Audio. Same qualities as the Twin unit, but is bus powered as well as Thunderbolt 3/USB C. So you will need this port.
MOTU M2	M2	$170	High class preamps and converters at a beginner's cost. This unit has one of the highest dynamic ranges of the desktop interfaces. Unfortunately the preamps can sometimes be more noisy than others.
Zoom UAC-2	UAC-2	$250	Zoom is a trusted company when it comes to field recording, so putting out a desktop interface was a smart move to make.

Solid State Logic 2	SSL 2	$230	SSL has been a trusted manufacturer in high end recording technology for over 40 years. Now they've released a high end 2 channel interface at an affordable cost. Simple to use, high dynamic range, and very quiet preamps.
Warm Audio WA12	WA 12 Mk II	$450	The tone is fantastic and rounds out the voice in a pleasing way. A great step up from a scarlett that emulates a classic sound. This is NOT a new interface, but an external preamp that you will then LINE IN to the interface.

These are interfaces that we can vouch for, but there are definitely some out there that we are not aware of that may be even better. If we are able to test additional interfaces and would vouch for them, we will not hesitate to update this list.

((DISCLAIMER:

The following subsection contains opinions based on experience and individual preferences. You have been warned. So don't @ us…

There are plenty of interfaces out there in the market that, while easy to use, are lacking in the quality department and are easily discernible from more quality interfaces. One of the most famous examples is the Focusrite Scarlett series. To clarify, we do not believe the Focusrite Scarlett series are *bad interfaces.* They are more than suitable for auditions and home-level productions. However, we believe that the Focusrite Scarlett series tends to cap out at that level due to the quality of the technology within the actual interface.

Having a Focusrite Scarlett interface DOES NOT AUTOMATICALLY DISMISS YOU from studio opportunities. Many of us have booked remote gigs with that setup or worse. We are recommending better interfaces to make sure that we raise the STANDARDIZED FLOOR of your audio quality (aka making sure the worst link amongst the community is still pretty good). Just because it worked a few times, doesn't mean it will always work. We are trying to INCREASE YOUR ODDS as much as possible so you can WORK AS MUCH AS POSSIBLE.))

Part 3: Digital Audio Workstation (DAW)

This list will be of typical DAWs that come to mind when you think of recording as well as possible choices for you to use at home, for WFH or auditioning.

DAW	SYSTEMS	DESCRIPTION
Pro Tools (First / Native / HD)	Mac & PC	Industry standard in recording software. PT First- FREE version of Pro Tools, basic recording and editing functionality. Sessions are saved on Avid's cloud service via Avid Link. Viable for WFH setups. NO VIDEO FUNCTIONALITY PT Native- Also known as just "Pro Tools." Used by amateurs and professionals. PT HD/Ultimate- Systems using HD software use external hardware as the computer's processing power. Thus enabling a larger work load.
Logic	Mac	Used by music producers, but has video functionality. Which will make WFH dubbing easier. Studios cannot send sessions, so you will have to take files and create your own when necessary.
Reaper	Mac & PC	Low cost / free multi track DAW. Has video functionality and is pretty user friendly. Giving out free trial licenses until June!
Twisted Wave	Mac	Easy to use, high functional, and affordable! Looks like Audacity with better post production qualities.
Adobe Audition	Mac & PC	Very intuitive, simple to use Lots of presets made for ease of use to clean up/prepare your audio for delivery "Auto Heal" for fixing pops/clicks/mouth noises

		Easy to use noise removal, for quick clean ups “Spectral Frequency Display” to show you which frequencies may be standing out in your recordings. Good to see noise, or buzz in the low end.
Audacity	Mac, PC & Linux	Free and user friendly. Typically an actor’s first DAW.
Garage Band	Mac	Basic multi track recorder made for iOS and OSX.

Part 4: The Microphone

A microphone is a device that takes audio (sound) and converts it into an electrical signal. From there, the signal is taken through a preamp, and from the preamp to an interface. The two main types of microphones that you as an actor will normally come across are CONDENSER mics and DYNAMIC mics.

TYPE	DESCRIPTION
Condenser	A wider frequency range and more 'natural frequency response' that will "make things sound how they are supposed to sound." Phantom power (48v) is needed to power the condenser mics in order to function properly. Acoustic treatment in room is needed for a clean sound
Dynamic	Better isolating to the voice in more noisy environments Sonically, has less dynamic range (can sound "compressed") Requires a different mic technique. You are able to stand much closer without worrying about overloading the capsule.

Example Microphones:

In no particular order, these are examples of the most widely used microphones in professional and home settings.

MICROPHONE	MIC TYPE	DESCRIPTION
Neumann u87	Condenser	Widely used in music and voice over. Amazing clarity and presence. Most expensive. ($3,200-$3,500 USD)
Neumann TLM 103	Condenser	Also widely used in voice over. Flat frequency response with a decent bump at 5Khz, meaning more presence (A more pronounced sound) $1,100 USD

Sennheiser MKH 416	Condenser	Known as a ‘shotgun’ mic, these are highly directional and must be pointed directly at the sound source. These mics require a larger room that is very well treated, otherwise, the mic will pick up other ambient noises because of the higher amount of gain used. $1,000 USD
Warm Audio WA-87	Condenser	High quality at an affordable rate. This mic is designed from high end, studio standard parts. Bright, present tones with a full body. $599 USD
Rode NT1	Condenser	Most affordable of microphones, but is a long standing workhorse microphone used in music and voice work. One of the quietest mics out there, and with one of the larger frequency responses, there is a reason this mic has been a go-to for professional and at home productions. $269 USD
Studio Projects C1	Condenser	High end quality while also boasting low noise. It is incredibly clear sounding. So much so that it is often referred to as a clone of the u87. Comes with built in low pass and pad features. $250 USD
Shure SM7B	Dynamic	This is a dynamic mic, so no phantom power is needed Widely used in radio and broadcast audio, this mic is used in studios all over the world in both voice work and music. Very small proximity range, you cannot deviate far from the front of the mic or else you will get a more off axis signal. This mic requires a stronger preamp for proper recordings, baseline products will not work. $400 USD

The Recording Process

Part 1: Microphone Placement

- This is going to be simple…
  - You should be between 10-16 inches away from the mic
  - There are times you can move closer or farther away
    - Shouting: You may want to take UP TO a step back so your voice does not overload the diaphragm of the mic, causing your sound to distort.
      - Your engineer will most likely remind you of this.
      - When working at home, you may need to experiment to see where you can stand, and where your preamp levels should be at for good recordings.
    - Whispering: Obviously because your baseline speaking level has gone down, you can move in a bit closer to the microphone to make sure you can be heard.
    - Both of these instances have different varying levels, be mindful how much deviating from your normal spot you do.

Part 2: Gain Control

We're not asking you to become engineers… but there are skills you may need to learn to adapt to all this home recording. We understand that this may be difficult to adapt to and we will try to keep that as simple as possible. We believe in you.

- **Recording**
  - Now that you are self recording, you will need to pay a bit more attention to your recording levels. It may seem self explanatory, but there are more times than not that audition files are sent with an abundance of normalization or other post processing. When recording your actual takes, you want a 'strong' and 'healthy' recording signal.
  - You are not expected to be recording close to or at 0db, but you also do not want to record as low as -20db and normalizing it up to proper level. (We can tell.)
  - Try to record somewhere in the middle while leaving yourself enough headroom (so that you don't peak). <u>Engineers typically try to aim to have -12db at minimum.</u>

- **The Gain Knob**
  - You will be interacting with this little guy more than you ever have. LEVEL CHECKS are done before each session because each actor has different dynamics. You should check your recording level before each session to make sure your levels are where they should be.
    - Reading as your character in a 'normal tone' of voice will decide what your regular recording level will be. (If you are always loud or high energy, then start there, same if you are quiet, or average)
  - Whenever you know your character will be LOUDER… Be mindful, and remember to lower your gain slightly as well as maybe take a half step back.
    - The same, yet opposite procedure goes for when you get more QUIET.
  - Once your dynamics come back down and level out, you must remember where your starting point was on the knob. This will preserve your original recording level.
  - You want to be able to give the best files you can WITHOUT NORMALIZATION.

Part 3: Don’t Touch Your Files

Hardware/Software Processing

“I keep hearing about compression/limiting/normalizing… What is it? Should I do it? How much should I do it?”
The short answer? No. Don’t do it.

When recording files that are to be delivered, you default thought is that they should be delivered RAW. Meaning no processing done whatsoever before, during, or after recording.

The processing will be handled by your edit and/or mix engineer.

Types of Processing

- Compression. Whenever audio is being recorded, engineers will usually ask their client… “To compress, or not to compress?”
    - Compression LOWERS “Dynamic range.” Bluntly meaning, the softer sounds are louder, and the louder sounds are softened. So everything is a more EQUAL level.
    - If you are sending audio to someone else to edit. Your best option is to NOT COMPRESS. Be mindful of your levels and all of the post processing will be taken care of by the editing engineer.

- Limiters
    - As its name implies, a limiter LIMITS how loud a signal can get. Keeping your files from peaking.
        - “This is good, right??? Then I won’t be sending anything that is peaking and is also LOUD?”
            - NO, while it is good that your file is loud and not peaking, a Limiter is like a hard compressor. Taking out any dynamic range if it crosses the threshold.

- Normalizing
    - Normalizing audio takes the LOUDEST point in your recording and brings it up to however loud you set it.
        - A negative side to normalizing, is if you have LOW LEVELS in your recording, then normalize them to HIGH LEVELS. You will bring up your NOISE FLOOR. Mostly known as that annoying HISS you all hate and try to avoid.
        - That is why getting a solid baseline recording level is important. And why have baseline equipment just won’t do it for Broadcast quality. They do not have enough power and quality most times to provide what studios need.

The Delivery

Part 1: Mono vs. Stereo

Mono, or monophonic sound reproduction, is intended to be heard as if it were a single channel of sound perceived as coming from one position.

Stereo, or Stereophonic sound, is a method of sound reproduction that creates an illusion of multi-directional audible perspective. Or more than one channel of sound perceived from more than one position.

Mono = 1 source

Stereo = Multiple sources (Usually 2 sources in most occasions)

As an actor, you project a single signal. 1 Mouth - 1 Signal. So while recording, you should only be recording to a mono track. AND, only providing a mono file. Unless specified to give stereo, which we highly doubt would happen, always send a mono file.

Part 2: .wav vs .mp3

A WAV file is an audio file that is uncompressed. Meaning that the file is being produced without any loss in quality. This allows us to listen to audio in the highest quality with a high dynamic range. Most DAWS will record audio in WAV, and that can later be converted to other formats. This type of file is used for “Broadcast Quality”.

An mp3 is an audio file that is VERY compressed. Meaning the file is lower in quality, and has less dynamic range. When converting to mp3, the frequencies that humans typically are unable to hear are removed, thus creating a smaller, more compressed file. While listening on normal headphones or speakers you may not be able to tell the difference between audio qualities. This type of file is typically used for sending auditions.

Part 3: Bit Depth/Sample Rate

A Sample Rate is the measurement of the number of Samples per second of audio. The more samples, the more information can be captured, the higher the quality. We typically record games and dubbing around a 48kHz Sample Rate.

Bit Depth is the amount of information packed into each Sample. This correlates directly to the Dynamic Range in audio. The higher the Bit Depth, the higher the Dynamic Range. A CD will have 16 bit audio while something like an iTunes mp3 will have 24 bit audio.

More than likely a studio will be recording at a 48kHz Sample Rate with a 24 Bit Depth. We can convert to a lower quality if necessary, but cannot convert higher without potential audio problems.

Part 4: File Naming

So this may be simple for a lot of you but we wanted to at least give a quick moment to this since you may be providing more than audition files. Whoever is sending out the work to you should be providing a template, but just in case we would prefer that you have a couple things in your files.

Your Name / Your Character's Name / The Project Codename or Title / Episode Number

They don't have to be in that order, but giving more information in each file helps us keep track. Source Connect (or something of the like) will more than likely be a thing in the future, but there is no guarantee they won't ask for files from your end as well. Be prepared to record from your end while they are recording you remotely.

<u>Terminology</u>

ADR: Automated Dialogue Replacement. AKA Dubbing to Timecoded video.
(Widely said to mean “Additional Dialogue Recording,” but that’s wrong)

Timecode: A sequence of numbers used for synchronization of different materials. Specific timecodes are marked for notable points in video. [HOURS:MINUTES:SECONDS:FRAMES]

Locked to picture: Pretty much a fancy way of saying that a system can run timecode synced with a video.

Editorial: When an engineer plays back files and takes out mouth noises, plosives, etc.…

Rolling: What engineers say when we are recording. Referencing tape recording. Because tape looks like rolling wheels I guess? Fuck if i know at this point.

Sibilants: Consonants that produce higher frequencies. Specifically when air moves outward through or towards the teeth.

Plosives: a consonant sound that is made by stopping air flowing out of the mouth, and then suddenly releasing it

Cans: Headphones. Also. Get your own personal pair. Do you want to share ear germs????

Wild Recording: When there is no picture or timing to record to. As an actor you are pretty much free to go when ready. No Beeps.

You may end up being asked to do wild takes when dubbing right after your initial take while the video and audio are still rolling. Just give another read and the engineer will edit into place.
Chasing: When you hear a line, mainly being the source material, then immediately follow up by a repeat or continuation of the line.