Home Studio Music Recording

George Williams — October 2015

In an article in the HTMA Newsletter, I gave you an outline of what I'll be talking about in this paper. I'll go into more depth on what you'll need to know to get you off the ground with home studio music recording.

1. Hardware

You'll need microphones, microphone stands, some kind of interface to your computer, cables, and a computer. You'll also need some headphones and/or speakers. Let's deal with these one at a time.

1.1. Microphones

Microphones ("mics" as I will refer to them) are something that you can spend a lot of money on, and there are many kinds to choose from. For HTMA's live performances, we use Sure SM-58 (vocal mics) and SM-57 (instrument mics), both of which cost about $100 each, and that is what I chose to use for my home recording. But there are many other kinds that you should consider, based upon your needs and budget. Here are some pointers to useful web resources:

- Microphones: Dynamic vs Ribbon vs Condensor: An overview of common microphone design and use in the studio
  http://soundschematic.com/dynamic-ribbon-condensor/
- A Beginner's Introduction to Microphone Polar Patterns
  http://ehomerecordingstudio.com/microphone-polar-patterns/
- The 6 Best Dynamic Microphones of ALL-TIME
  http://ehomerecordingstudio.com/dynamic-studio-microphone/
- The 7 Best Microphones for Recording Vocals: under $700
  http://ehomerecordingstudio.com/best-vocal-mics/
- The 7 Best Ribbon Mics for Home Recording
  http://ehomerecordingstudio.com/best-ribbon-mics/

I recommend you get (at least) one microphone for each sound source. That is one for each vocalist, and at least one for each instrument. I say "at least" because sometimes it's desirable to have multiple mics pointing at different parts of an instrument. For example, different parts of a guitar produce sound with different qualities. I'll say more about that in a later section, but to start with, one for each sound source will be enough.

Some instruments have built-in electronics to capture their sound, and some of these do a very good job. But I think you'll find that most of these systems produce a different sound quality from an acoustic instrument with a separate mic. One may or may not be better than the other, they are just different. If you have an acoustic instrument with built-in electronics, I suggest you record both ways, and see what they sound like later when you're editing your music. You may even want to include both in your mix.

1.2. Microphone Stands

Microphone stands are necessary to hold your microphones in the proper position to capture the sound from your voices and your instruments. I'll talk more about how to position your mics in a later section, but I think you'll find that you want a stand that gives you a lot of flexibility in where you put your mic, while keeping the stand as much out of your way as possible. That means you probably want a *boom* stand, that is a stand with a vertical component plus a boom that holds the mic on the end and can be oriented in various ways. Here are some good options:

- On Stage Stands MS7701B Tripod Boom Microphone Stand
  http://www.amazon.com/Stage-Stands-MS7701B-Tripod-Microphone/dp/B000978D58
- On Stage MS9701TBPLUS Platinum Series Tele-Boom Microphone Stand
  http://www.amazon.com/Stage-MS9701TBPLUS-Platinum-Tele-Boom-Microphone/dp/B00TY3OQY0
- DR Pro Tripod Mic Stand with Telescoping Boom
  http://www.amazon.com/DR-Pro-Tripod-Stand-Telescoping/dp/B003G45V1U

Depending on how many mics and stands you need, you can sometimes find packages that include multiple stands, cables, mic clips, etc. Speaking of clips, depending on the kind of mic and stand you get, you'll need a clip to attach the mic to the stand. Many mics, such as the SM-57/58s, come with clips, but be sure to get clips that will fit your particular mics.

1.3. Computer Interface

The interface between your sound sources (microphones and/or instruments) and your computer performs a number of important functions. It takes signals from your various sound sources, amplifies the signal as necessary depending on the kind of input device, allows you to adjust the sound levels of each input to get the best signal from each source, and converts each of those signals into a digital form that your computer can use. Some interfaces also handle *MIDI* (Musical Instrument Digital Interface, https://en.wikipedia.org/wiki/MIDI) devices, which are beyond the scope of this discussion, and many interfaces also provide output channels from your computer for headphones, speakers, etc.

Notice that there is no need for a mixer. The DAW software (as I'll discuss in a later section) will perform this function.

Also, some sound sources, notably condenser microphones, require power for them to work (this is typically called *phantom power*, https://en.wikipedia.org/wiki/Phantom_power); whatever you plug such sound sources into must supply this power. So if your sound source needs phantom power, be sure that your interface provides it. (Note: phantom power is different from so-called T-power, which is common for portable applications. See the following for more information:
http://stason.org/TULARC/entertainment/audio/pro/3-5-What-is-phantom-power-What-is-T-power.html)

It's very important to understand that different sound sources produce different signal levels (voltages or power, depending on the spec). There are three (or four, depending on how you count them) different signal levels:

1. *Mic level* (-56 to -40 dBm, or around 1.5 mv to around 70 mv)
2. *Line level*, of which there are two kinds:
   - *consumer* line level is at -10dBV
   - *professional* line level is at +4dBu (or dBm)
3. *Instrument level*, which varies depending on the instrument, and is generally somewhere between mic and line levels

For more detail, take a look at the following:

Mic, Line and Instrument Level – What's the Difference?
http://recordmixandmaster.com/2010-02-mic-line-and-instrument-level-whats-the-difference

Technical aside: The units (dBm, dBV, dBu, etc) for measuring signal levels can be very confusing. Some of these units specify voltage, others power, and the relationship depends on the impedance of the items in the circuit (such as instruments or microphones). Some sources are "high" impedance, and some are "low." For those wanting some explanation, you could look at the following:

- Microphones: High vs. Low Impedance
  http://www.datalinkplus.com/procomm/BASICTR/micimpdc.htm
- dBV, dBu, and dBm | discrete music
  https://discretemusic.wordpress.com/2009/06/04/dbv-dbu-and-dbm/

The important thing to understand is that your computer interface must be able to accommodate the output levels of your various sound sources. For example, the interface that I have (a Scarlett 18i8, http://us.focusrite.com/usb-audio-interfaces/scarlett-18i8) can handle up to four mic-level sources, two instrument level sources, and up to eight line level sources, though not all at the same time because some of the connectors perform multiple duties.

The other side of your computer interface is the one that plugs into your computer, and there are several different kinds, USB and FireWire being the most common. Most computers don't have FireWire, so I suggest you look for one with a USB connection to your computer. Specifically you want USB 2, which is available on most relatively recent computers, and is fast enough to handle the data rate necessary to record as many separate tracks as you will likely have in a home studio environment.

How should you choose an interface? Look for one that can handle the number and kinds of inputs and outputs you want, and look at the specifications and reviews. Focus especially on comments about the quality of the amplifiers (you want low noise levels), crosstalk between channels (you want as little as possible), the quality of the A/D (audio to digital) converters, and make sure the interface is compatible with the operating system on your computer. Here is one of many sites you can look at for some ideas:

The Computer Audio Interface Guide
http://www.silentway.com/interfaces

1.4. Cables

Microphone and instrument cables are necessary to connect your sound sources to the interface to your computer. You'll notice that there are several different kinds of connectors and cables. The most common are:

- Balanced cables with XLR connectors, with a male connector on one end, and a female connector on the other end. These are typically used for microphones.
- Unbalanced cables with 1/4" (diameter) connectors (sometimes called QIC, for quarter-inch connectors), which typically have a male connector on each end. These are typically used for instruments, such as guitars.

There are some others that you may encounter for other applications, but those are beyond the scope of this paper.

There is often a good deal of debate about how much money to spend on cables, with some people saying you should get the best ones you can afford, often citing things like gold-plated contacts. I'll just suggest that you should look at reviews, and get ones with reviews where people don't complain about high impedance, noise, or poor reliability.

1.5. Your Computer

You'll need a computer to do the recording and editing. Most any kind of personal computer (Windows or Mac) will probably do, but I recommend a laptop. One reason is that you can more easily take a laptop to wherever you will be recording, while desktop models are less portable. Another reason is that laptops tend to be quieter than most desktop computers, which often have fans that can be noisy. And you don't need a real powerhouse of a computer; the performance demands for audio processing software are generally not very extreme.

If you have a laptop that has a fan that comes on when it gets hot, and it makes very much noise, this can be a problem. For this reason, I recommend a laptop that isn't a power-house model, at least for recording, since the less powerful laptops tend not to have such noisy fans. (I actually use a laptop for recording, then move the files to a desktop for editing, but that is more a matter of convenience and personal preference than necessity.)

1.6. Speakers and headphones

Speakers and/or headphones are essential to hear what you've recorded. Most computers today have some kind of speakers, but they're usually pretty lousy for listening closely to music. You can get studio-quality speakers, but they're expensive, and a lot depends on the acoustic qualities of the room they're in.

In my opinion, good headphones are actually more important than speakers, for a couple of reasons:

- Decent headphones are considerably cheaper (and better, really) than good speakers.
- When you are recording multiple takes, you will generally want to listen to other tracks and earlier takes. I'll say more on that later.

How should you choose your headphones? There are at least three important things to think about:

1. Sound quality,
2. Sound isolation, and
3. Comfort

Sound quality is obviously important when you'll be using these headphones to listen critically to your recorded sound when you are editing it.

Sound isolation is important for two reasons:

1. When you are editing and mixing, if there is a lot of ambient sound around you and it leaks into what you hear, then it will be hard to evaluate the small tweaks you may be trying to perfect.
2. When recording a new take, you will generally want to be listening to an earlier recording/take. If the sound leaks from the headphones out to where your microphone can pick it up, that's something you'll never be able to edit out of your recording.

Comfort is pretty important if you're going to be wearing the headphones for long periods of time — like when you're editing! But different people find different kinds of headphones comfortable or uncomfortable, so perhaps you should try wearing different kinds of headphones for a while to see what you like or dislike. One possibility: consider different kinds of headphones for the two kinds of usage:

1. When editing and mixing, sound quality is very important, but so is comfort. Sound isolation may be less important, depending on the ambient noise level where you'll be editing.
2. When monitoring another recording while recording a new take, sound quality isn't as important as when you're editing. Also, you won't be wearing the headphones for so long, so comfort may not be as important. But sound isolation will be very important.

To get some tips on buying headphones, just google something like "buying headphones guide" and you'll find lots of advice and options.

2. Software

In this section I'll talk about the DAW (digital audio workstation) software, including the one I use and recommend, and some important capabilities you should expect to find in the DAW you choose. I'll tell you which DAW I use, and why.

2.1. Digital Audio Workstation (DAW) Software

As with any powerful tool, there is a substantial learning curve. This is true of all of the comparable DAW packages I have encountered. There should be good documentation, and hopefully some tutorials available online, a selection of plugins, an active user community, etc.

No matter which of the many DAW packages you choose, they should all have a number of capabilities in common:

- Recording
- Editing
- Normalizing
- Mixing
- Busses and routing
- Plugins and other extensions
- Automation

(This is not intended to be a complete list. These are just some of the important ones that I want to talk about in this paper.)

2.1.1. Recording

You will use your DAW to record the sound coming from your various sound sources, through your computer interface, into your computer and DAW.

You will need to configure your computer and interface to talk to each other. This process is beyond the scope of this paper; see the documentation for your computer interface. But once that is done, you should be able to use your DAW to setup a project to record one or more tracks, one for each of your sound sources. You will need to specify which channel from your interface (associated with a single sound source) is to be associated with each track. Once that is done, you will be able to record your music into the various tracks of your project.

You will most likely want to record additional takes of one or more of your tracks. While you are recording new takes, or new tracks to be mixed with other previously recorded tracks, you'll want to listen to the earlier recordings for the following reasons:

- To assure that everything is recorded with the same tempo, especially if there are tempo changes, and
- Using the DAW to both playback the earlier recording and record the new take makes it possible for the DAW to synchronize the recordings in your project.

With most DAWs, you can record a new take for just part of a track. You can later use your DAW to specify which take is to be used, for any part of the track, when you play it back. You can even switch back and forth between takes, at any point in the track.

2.1.2. Editing

Once you have recordings on your computer, you can use your DAW to edit them in various ways. For instance, when recording a take, there is typically some silence or discussion among the performers before and after the track begins and ends. So one of the first things I usually do is trim that stuff out so that the track begins right at the beginning of the track, and ends right at the end.

You can cut or copy segments of your recording, and paste them somewhere else in your project, just like you can cut and paste in a word processor. For example, you could decide to change the order of two verses (or any other segment of your recording). Or you could decide to use part of one take, and part of another, on a per-track basis. Most DAWs let you zoom your view of the waveforms (audio signals) in or out time-wise, so this allows you to position your edits very precisely.

When you cut and paste sound segments, or switch between takes in mid-tune, there would normally be an audible transition. The way to fix this is by fading out the previous one while fading in the subsequent one, with some overlap between the two where the fade in/out takes place; this is called *cross-fading*. Most DAWs provide some support to simplify making this kind of transition as seamless as possible, with many giving you control of the duration and shape of the fade in/out profiles.
Fade (audio engineering)
https://en.wikipedia.org/wiki/Fade_(audio_engineering)#Crossfading

Most DAWs will also let you change the tempo of part (or all) of a recording. Changing tempo by compressing or expanding music time-wise would normally affect its pitch, but most DAWs can optionally prevent this. Such pitch control also means that most DAWs can also change pitch without affecting tempo. Neat trick!

2.1.3. Normalizing

The term *normalizing* refers to adjusting the volume of an item to a standard level, usually so that it is no louder than 0 dB[1] (or just below that level), and all DAWs should provide this capability. Because of the differences in how each sound source is mic'd and recorded, the volume levels for the various files are often different. Additionally, the volume levels typically peak well below 0 dB, which is the maximum level that you want to use; anything above that level will get cut off or distorted. To simplify the process of mixing all your tracks together, you will want to adjust the volume levels so that they all have a peak volume near 0 dB. This is the first thing I usually do after I've done the first editing of all the tracks, before I begin mixing. (However, I've recently read "It's not

1 Throughout these articles, when I use *dB* without qualification (as opposed to dBm, dBV, dBu, etc.), I mean dBFS, for decibels relative to *Full Scale*. See:
https://en.wikipedia.org/wiki/DBFS

necessarily a good idea to add normalisation, as that means another stage of DSP (which may degrade the sound, however slightly) — and you may need to change the overall level anyway when assembling all the mixes into a finished album."[1] I'll say more about mastering in a later section.)

2.1.4. Mixing

Mixing is the process of combining your various tracks in a pleasing way. You will want to adjust the volume of each track relative to the others so that they fit together the way you want them to. You can also *pan* tracks, to place different sound sources where you want them in the stereo field. You can also adjust the stereo *width* to control how localized a track is within the stereo field, and add *depth* using special effects, such as reverb (more on that later).

- Using the Stereo Field in Home Recording
  http://www.dummies.com/how-to/content/using-the-stereo-field-in-home-recording.html
- Improving Your Stereo Mixing
  http://www.soundonsound.com/sos/oct00/articles/stereomix.htm

You can start with just a few tracks enabled to begin with, or you can enable them all, and adjust things as you feel the need to. The process and order of doing these things is up to you, and you may want to do it differently on different projects.

An important part of this process is adjusting the volume levels so that your combined volume level for all your tracks (as combined into your master track) never goes above 0 dB. Most DAWs will provide a red light on each track that lights up when this happens.

2.1.5. Busses and Routing

Most DAWs also allow you to create *busses*, which are a way to collect the sound from other tracks (or busses) so that you can control them as a group. For example, you might want to group all your vocal tracks into one bus, and all your instruments into another bus. Once you adjust the volume levels of all your vocal parts relative to each other, and do similarly for your instruments, you can then adjust the volume of all your vocals relative to all your instruments using just the volume controls of those two busses. Most anything you can do with a track, you can do with a bus. How you use busses is up to you, but they can be very handy.

Routing is the ability to control for each track or bus where a signal comes from and where it goes. This is essential for using busses, but it's also important for creating different output channels for use with speakers and headphones.

1 Audio Mastering In Your Computer:
http://www.soundonsound.com/sos/aug04/articles/computermastering.htm

2.1.6. Extensions, especially Plugins

Most DAW packages can be extended in various ways. DAWs often come with a selection of *plugins* that you can use to add various kinds of effects (*FX* is the common shorthand for this), including tone (typically called *EQ*, for equalization), reverb, echo, audio compression, pitch shifting, spectrum analysis, and a host of other things you can learn about on your own. There are many plugins available on the internet that you can get, some free and some not.

You can also put FX plugins on busses, so you can do things like use the same reverb settings for all your vocals. Putting common FX plugins on a bus also reduces the processing load on your computer over having the same processing done for each of several tracks. Plugin processing usually introduces delays into the playback, and your DAW needs to accommodate this to keep things synchronized across tracks. This delay can become noticeable when you are moving the playback start point around, or when you are adding another take while listening to an earlier take. Generally, you shouldn't have to worry about this though.

Most DAWs support other ways to extend them besides plugins. One example is building a *macro*, which is just a way to repeat a series of common steps. Many DAWs also provide scripting mechanisms, and/or ways for programmers to write code that can be loaded into the DAW to make it do things that aren't built into it by the DAW developers. Often some really nifty extensions are available free or for only a nominal fee. Check your DAW's documentation and website.

2.1.7. Automation

Another feature of most DAWs is typically called automation. Here's an example of when you might need this capability: Say you have one player who has played a note or a chord on his instrument a bit louder than normal every once in a while. You could use an audio level compression plugin to reduce the dynamic range on this track, but although compression is a very useful tool, it may not be the best way to handle this occasional situation.

- Dynamic range compression
  https://en.wikipedia.org/wiki/Dynamic_range_compression
- Compressing Acoustic Guitar
  https://www.homemusicstudio1.com/compressing-acoustic-guitar/

An alternative is to use automation to lower the volume of the offending track briefly, just enough to even out the sound of the instrument overall. You can also use automation to control other things, like panning, stereo width, muting, controlling your FX (including turning them on and off, and controlling their settings).

Different DAWs implement this in different ways, and sometimes there are several ways to use it within a single DAW; I'll leave it to you to discover how to do this within your DAW. But automation is often very useful. You'll want to learn about this.

2.2. A Recommendation

There are many options and opinions, and there are many aspects of using DAW software, some of which I will touch on later. But the one I use is *Reaper:*

- REAPER | Audio Production Without Limits
  http://www.reaper.fm
- REAPER
  https://en.wikipedia.org/wiki/REAPER
- Ep 27 | Reaper DAW – 7 Reasons I Record With It
  https://www.homemusicstudio1.com/reaper-daw-ep-27/

Based on my relatively modest experience, this is the best option I've encountered for the home recording environment. It is an incredibly flexible and powerful tool for the entire spectrum of home studio sound production. It will run on almost any reasonable computer (Windows and Mac), and yet it is very modestly priced: $60 for the non-professional user. And you can download a free, non-crippled version of it to evaluate.

Another option that you might want to consider is Mixcraft Pro Studio 7:

Mixcraft Pro Studio 7 Review
https://www.homemusicstudio1.com/mixcraft-pro-studio-7-review/

Both of the homemusicstudio1.com videos above give some good ideas of what's important in a DAW package, plus a look at what such software looks like.

3. Setting Up To Record

In this section, I want to talk about setting up to record, including the recording environment (where you will be recording), positioning your mics, and how to use your mics effectively.

3.1. Acoustic Treatment

In a professional recording studio, the rooms are often acoustically treated with two objectives:

1. eliminating exterior sounds from entering the room (*soundproofing*), and
2. controlling the reflection of sound within the room (*acoustic treatment*).

It's impossible to achieve either of these objectives completely, but they represent the ideal. It's important to understand the difference between these two objectives because you use different means to achieve them.

Strategies for reducing exterior sounds include things like building double walls, adding sound insulation materials to the walls, and other steps that can get quite expensive (none of which I have done). But it also includes relatively simple things like turning off the air conditioning/heating system while you're recording, putting the pets outside, or even choosing a time of day when the neighbor isn't mowing his lawn or shooting his pistol in his back yard (yes, that happens out in the county, where I live).

I'm not going to spend a lot of time discussing soundproofing in more detail, but if you want to know more, you might start by looking at the following:

How to Soundproof a Room for Music Recording
http://ehomerecordingstudio.com/soundproof-room/

Strategies for reducing sound reflections include both absorption and diffusion. Recording in a room with rugs or carpeting, and few large flat, reflective surfaces goes a long way. Actually, I haven't done any soundproofing or acoustic treatment where I record, and at least for the simple kinds of recording I do (two instruments and two vocalists, with fairly close mic placement), I haven't noticed any significant sound reflection problems in my recordings, but that may just be my relatively untrained ears.

For more aggressive acoustic treatment, professional studios will often use:

- *Bass traps* to absorb the low frequencies
- *Acoustic panels* to absorb the mid/high frequencies
- *Diffusers* to scatter the remaining frequencies

As with soundproofing, I'm not going to spend a lot of time discussing the details of what these things are or how to use them. But if you want to delve into this some more, you could start here:

Acoustic Treatment 101: Getting Your Room to Sound Great
http://ehomerecordingstudio.com/acoustic-treatment-101/

3.2. Arranging Your Room

Next you will want to arrange everything where you're going to record. This includes:

- The *artists:* vocalists and instrumentalists, including their instruments and music stands, etc.
- The *recording gear:* microphones; computer; preamps; cabling; etc.

Typically, the artists want to be able to see each other while recording, so that's a factor. And if you (the reader) are both an artist in the recording, and operating the recording equipment, configuring everything around you can be challenging. Much depends on how many people there are, what kinds of instruments are involved, and how much space you have. I don't have any general advice, but expect to spend some time figuring out how to arrange all these things in the space you have available.

If you have the space and time, try recording with each sound source in different parts of the room. Placement can make a big difference in the quality and nature of the sound that you record. Experiment, and you may be pleasantly surprised at the results.

A critical part of configuring your room is where to put the mic stands, and how to point the mics. Ideally, you want each mic to record one, and only one, sound source. To achieve this, you will want the mics...

- to be as far away from other sound sources as possible,
- within a few inches (I'd say 3 to 6 inches at most, if possible) of the sound source each is intended to record, and
- pointing away from other sound sources, as much as possible.

3.3. Using Microphones

Most vocalists need to *learn how* to sing into a microphone — good mic technique doesn't come naturally. Many vocalists move their heads around when singing, which causes the volume and tonal quality (as captured by the microphone) to change in the recording. Jerry LeCroy created a presentation that covers some of the key ideas, and his presentation can be found on the HTMA website at:

http://www.huntsvillefolk.org/articles.html

If there are passages where the vocalist sings much louder than others, they need to learn to move back from the mic a bit during these passages. Putting the vocalist in headphones, so they can hear what the mic picks up often helps them to learn what they need to do to get the sound they want.

Also, if a vocalist gets within an inch or so of the mic (depending on the type of mic), there is a *proximity effect* that causes the mic to increase its low frequency response. This effect can be used to advantage in some cases, but may be undesirable in others. See also:

- Proximity effect (audio)
  https://en.wikipedia.org/wiki/Proximity_effect_(audio)

- Microphone Proximity Effect
  http://www.tangible-technology.com/microphones/proximity/proximity.htm

Pointing mics at your instruments is a bit of an art. For example, with acoustic guitars, you don't want to point the mic at the sound hole — this will produce an unpleasant "boomy" sound. Depending on the instrument, performer, type of music, etc., you often want to point a mic somewhere around the 12th or 14th fret, possibly pointed a bit toward the body of the guitar. Or you may want to point one mic more at the body of the guitar, and another a bit farther up the neck, though with multiple mics you have to be careful not to introduce phase anomalies (where the signal from one mic is out of phase with the other, canceling each other out). These different placements will get different aspects of the sound produced by the instrument. Multiple mics can also be useful for instruments (or vocalists) who move around when performing; try to position them such that when the sound source moves out of the range of one mic, it moves into the range of another. This is another part of your setup where it can pay big to experiment.

Other instruments require different mic positioning and pointing techniques. Fiddles, hammered and mountain (lap) dulcimers seem to be simpler — just point at the instrument from above. For steel drums, you probably want a mic under each drum. Pianos are very complex instruments, and may require as many as five mics. And micing drum kits (like you find in most rock bands) is a complex art unto itself, very difficult to handle in a home studio.

- Mic Your Acoustic Guitar Like a Boss
  http://www.jamplay.com/weekend-warrior/w/mic-your-acoustic-like-a-boss
- The Recording Guitarist: Mic Makes Right
  http://www.premierguitar.com/articles/20356-the-recording-guitarist-mic-makes-right
- Glyn Johns drum mic setup
  http://danalexanderaudio.com/glynjohns.htm

If you have a sibilant vocalist (one with noticeable hissing, "ess" or "shh" sounds), or one whose breath sometimes makes a popping sound in the mic, try pointing the microphone up at about a 45 degree angle, and you might find much of the popping and hissing goes away. Not all vocalists need them, but for some, you may also want a pop shield, such as:

Nady MPF-6 6-Inch Clamp On Microphone Pop Filter with Flexible Gooseneck and Metal Stabilizing Arm
http://www.amazon.com/Nady-MPF-6-Microphone-Gooseneck-Stabilizing/dp/B0002CZW0Y/

3.4. Using Your DAW to Record

Before you actually start recording, you'll want to *adjust the gains* (volume levels) on each of your input signals to be as strong as possible, but without overdriving either your preamps or the input channels of your computer interface. Most preamps and interfaces will have either meters or red lights to tell you when you have things turned up too high. The volume levels will likely be different for different sound sources. I recommend you actually sing/play one or two pieces where each of the performers has a passage that is as loud as they are likely to get, watching your indicators, and adjusting the levels as necessary.

If you have some instruments or vocalists who are rather loud, you may find that the sound they produce will be picked up by some of the other mics (those intended for other vocalists or instrumentalists). If this happens, you may want to lay down an initial recording, possibly with only a subset of your performing group. Then, *record additional takes*, one performer at a time, while that performer listens to other performers or another take through headphones. You may or may not discard the original take, but this technique will yield relatively clean recordings of each performer, which can then be edited and mixed in your DAW software on your computer. This technique assures that all the takes are recorded with consistent tempo, making it much easier to mix them in the DAW.

Also, it is important for the signal path from the microphone to the computer which performs the recording (including cables, preamps, etc.) to be kept as simple as possible. Every box, cable, and connector in that signal path is another place where noise and distortion can be introduced. There's no need to add effects (tone, compression, reverb, etc.), because all that can be done in the DAW, and that's the place where it's easiest to control. And anything that's done to the signal before it's recorded can't be undone in the DAW. While recording, keep it simple.

4. Post-Production

In previous sections, I've already talked about making the recording (including multiple takes), which is typically called the *production* phase of a recording project. In this section, I want to talk a bit about post-production, especially focusing on sound design and mastering.

Audio post production is the general term for all stages of the project happening between completion of the initial recording and completion of a master recording. It involves sound editing, sound mixing, sound design, and the addition of effects.
https://en.wikipedia.org/wiki/Audio_post_production

Since I've already talked a good bit about editing and mixing in the section about DAW software, I'm not going to say much about that here. Instead, I want to focus on sound design and mastering, which are two related but very different things. Both of these topics are often considered part of post-production, and are closely related to editing and mixing.

4.1. Sound Design

Sound design overlaps somewhat with other steps in the post production process. The general idea is to use your tools to create a mood, to help evoke certain emotions with your music. Sometimes you can achieve this by adding reverb, echo, or other FX, or adding certain kinds of instrumental or special effects in places, or changing which parts of your mix come to the fore. This is an art form in itself, and is mostly beyond my current levels of knowledge and skill, but it's something I want to know more about, and I wanted to at least mention it. Here's a Wikipedia article that talks about it:
https://en.wikipedia.org/wiki/Sound_design
(Skip down to the section titled "Music.")

4.2. Mastering

Mastering is sometimes considered to be part of the post production process, but you might be better off thinking of it as a separate step. This is partly because the rest of post production is performed on each musical piece individually, whereas mastering deals with all of the pieces as a group. It is about creating a collection that is polished, and feels like a collection. It is also about creating a final set of files from which all copies will be produced.
https://en.wikipedia.org/wiki/Audio_mastering

Mastering has an artistic aspect, helping to optimize the intent of the sound design, But among other things, mastering also is intended to produce recordings that all have roughly the same volume level, and will sound good on just about any kind of sound system, from a boom box or your car CD player, to a top-end surround sound system. This is very difficult to do in a home studio environment.

One of the best discussions I've found so far on mastering is here:
Audio Mastering In Your Computer
http://www.soundonsound.com/sos/aug04/articles/computermastering.htm

You can perform the mastering process yourself in your DAW, but if you want a truly professional level result, you should consider sending your projects to a specialist. However, if you're just interested in producing recordings for your own use, then you can produce very reasonable results yourself at home.

4.3. Additional Resources

I'm far from an expert at knowing what to do, and in what order. Probably the best advice I can give you at this point is to go spend some time watching some of the free videos available on youtube on the subject of home music production. One of the best sources I've found is Dave Maxey, of Home Music Studio 1:

- Home Music Studio 1 - YouTube
  https://www.youtube.com/user/hmusicstudio1
  https://www.youtube.com/user/hmusicstudio1/videos
- Ep 20 | Building a Professional Mix With the 4-3 Framework
  https://www.youtube.com/watch?v=NeI6ZSwMI8A
- Home Recording Tips for Pro Audio on a Budget | Home Music Studio 1 Podcast
  https://itunes.apple.com/us/podcast/home-recording-tips-for-pro/id541674753

Dave Maxey also offers lessons and tutoring/mentoring, for a fee (I have no relationship with Dave Maxey beyond the fact that I've watched/listened to a good bit of his material):

- Home Music Studio Recording
  https://www.homemusicstudio1.com/home-music-studio-recording/

(Dave can be somewhat wordy, and does a good bit of self-promotion in his free material, but if you bear with him, what he has to say is usually quite good.)

There are also a lot of other resources available on the web. I've learned a lot from using a search engine to find more information about topics that I've encountered in my reading. You can also learn a lot from experimenting and talking to other people who are exploring the art of recording and producing music. Always keep learning!

I hope I haven't scared you away from home studio recording. This is actually a lot of fun. I also hope you'll give me feedback on the topics I've tried to address in this paper. My email is:

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