The Podcaster's Audio Handbook

A Technical Guide for Creative People

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Corey Marie Green

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About the Author

Corey Marie Green is an audio engineer from Melbourne, Australia, who specializes in podcasting and radio. She has filled many roles in her radio career, including journalist, producer, editor, and sound engineer at live events. Through her business, Transducer Audio, she provides a range of services for podcasters including editing, content development, and training.

Photo Credit: Karoline Morwitzer

About the Technical Reviewer

After completing studies in both sound production and electronic engineering, **Riah Williams** has been professionally active in the communications, broadcast, and music industries. As Station Technical Worker at 3CR Community Radio in Melbourne, Riah has set up remote live broadcasts from rallies and blockades, parks, pubs, and inside prisons. In 2014, Riah designed talkback equipment which was installed in 16 community radio stations in Timor-Leste. Riah's time is currently divided between 3CR and repairing music instrument amplifiers at Open Ear Audio.

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Preface

UK comedian Deborah Frances-White was about to quit comedy due to the grind of dealing with sexism in the industry. Instead, she decided to start a podcast: *The Guilty Feminist*. When she shared her experiences, she found a community of like-minded people and created a much-needed space for diverse comedians.[1]

The audio quality of your podcast is important, because your podcast is important. Most podcasters start out because they're passionate about a topic and want to share that passion with other people. Your podcast might make people laugh, teach them a new skill, or inform them about an important issue. Podcasts are an excellent way to build community. However, poor-quality audio can be an impediment to reaching your audience.

The Importance of Audio Quality

Audio quality sends a message to your listeners about how much you value your ideas. And it's not just your ideas: if you invite a guest onto your podcast, you will want to give them the best platform possible. If you value your work, then it will encourage others to do the same.

Good-quality audio is not just an extra, it's an accessibility issue. Improving your audio quality will make your work available to a much wider audience.

[1] Frances-White, D., 2018. *The Guilty Feminist*: *From Our Noble Goals To Our Worst Hypocrisies*. London: Virago.

You should also consider that the audio you release into the world might not reach your listeners in the same state as you sent it. Even if your initial recording sounds OK, a podcast that has been played through a streaming service undergoes processing. On top of that, you have no control over the equipment your listeners are using or the circumstances in which they're listening. You do have control over the recording and editing process.

You can't stop a member of your audience from listening to your podcast in a laundromat out of an old shoe, but you can release a podcast with high-quality audio

High-Quality Audio Is Accessible Audio

Maximizing your audio quality makes your podcast accessible to a wider range of people. When creating a podcast, you should consider who your audience might be and what needs they have. Low-quality audio could make it impossible for certain people to access your work.

People who are hard of hearing particularly need a podcast to be clear. This might be a specific consideration if you expect your audience to be older or to be made up of people who frequent rock concerts.

Another thing to consider is that you or an interviewee might have difficulty speaking. Alternatively, you could be speaking with an accent that is different to your listener's. It's incredibly important to include a diversity of voices in the media, and independent media such as podcasts often take up this work. In this instance, a high-quality recording will be helpful for the listener.

Your listener's attention may be divided between your podcast and another task. In 2019, Edison Research surveyed podcast listeners in the United States and found that 59% of respondents had listened to a podcast while doing chores.[2] You may be competing for your listener's attention with the task of removing mold from between bathroom tiles. You will know from your own experience that it takes your full attention to be able to understand poor-quality audio. You want to make it easy for your listener to be able to concentrate on both tasks.

Often people listen to podcasts in noisy environments: on the train, while driving a car, or while working. Audio that has inconsistent volume levels is impossible to listen to in these circumstances.

Your listener might be using poor-quality equipment, or they might have a poor Internet connection. It's not uncommon for people to listen to a podcast using low-quality earbuds or out of the tinny speaker of their smartphone.

[2] Edison Research. 2019. *The Infinite Dial - The Podcast Consumer 2019*. [online] Available at: `www.edisonresearch.com/wp-content/uploads/2019/04/Edison-`Research-Podcast-Consumer-2019.pdf

Podcasts are often played through the speaker of a smartphone. This can significantly reduce audio quality

I have prepared an audio file so you can appreciate how much audio quality can degrade between your original work and the final playback. In this case, I have uploaded the file to the Internet, streamed it through Apple Podcasts, and played it back through the speakers of my smartphone on my parents' porch (Audio 0-1).

Audio 0-1 How your podcast might sound when played through the speakers of a smartphone in a noisy environment

You can hear quite a difference between the quality of the audio at the beginning of the file and the quality at the end. The words I am speaking are still clear at the end of the file, but only because the original recording was of a high quality.

If you do everything you can to maintain the quality of your audio files through the recording and editing process, you'll be making your podcast accessible to the widest audience.

Maximizing Audio Quality to Compensate for Challenging Conditions

One of the great things about podcasting is that if you live somewhere with an Internet connection, there's a low barrier to entry. That does not mean that podcasts are created on an even playing field.

Sometimes your recording environment can be quite noisy, or you might have to work with low-end equipment. You might need to conduct an interview over the phone or the Internet, which is convenient but can compromise audio quality. It is at these times that it is particularly important to understand audio engineering techniques.

Regardless of your circumstances, you can improve the quality of your podcast using the audio engineering techniques contained within this book.

Understanding audio engineering techniques will help you make a clear recording in a noisy environment, such as a UFO convention

PREFACE

How This Book Works

The information in this book will help you to optimize the audio quality of your podcast in the easiest and most effective ways. Once you have a handle on some audio basics, you can focus on expressing yourself creatively and making content that will connect with your audience.

This book is intended for both new podcasters and well-established podcasters who wish to improve their audio. For many people, podcasting is their first exposure to audio engineering. While there are many online resources available for podcasters who wish to improve their audio, it can be hard to know where to start. Many podcasters will make a few recordings, run into problems, and then look for a way to fix them. The solutions they find can be difficult or expensive and often unsatisfactory. This guide provides an overview, which will help you avoid common audio issues before they arise. I know that people often dip in and out of technical books, but there is a benefit to reading this book in chapter order.

Many online resources assume a high level of technical knowledge. My aim is to make the information in this guide accessible to people who don't have that background. As mentioned, I have concentrated on the simplest and most effective techniques for improving audio quality. I have presented the information in plain English and demonstrated my points with real-world experiences. I explain key audio terms as they arise, and you will also find a glossary. The most important points in every chapter are summarized so that you can refer to them later. This guide will provide you with a solid foundation in basic audio techniques: enough information to get started, make a really good recording, and shape it into a podcast. If you then decide you want to learn more about audio engineering as it relates to podcasting, then having this solid foundation will make it much easier for you to access audio industry resources.

One of the principles that I work from is that it's easier and more effective to make a high-quality recording than to fix a low-quality recording. As such, most of the book concentrates on the process of making a high-quality recording, with only two chapters on editing at the end. Even though a large part of my work as a sound engineer involves repairing audio, I barely cover this topic in the book because it gets quite complicated. You can think of fixing a bad recording as being a bit like trying to fix a cake after you've baked it. You can cut the burnt bits off the cake and add more icing, but if you really want a good cake, you'll need to start again – this time with a recipe book.

Trying to fix a bad recording can be a bit like cutting the burnt bits off a cake and adding more icing

It's easier to make a high-quality recording if you're using suitable audio equipment, but not everyone has a huge budget to spend on the latest gear. Where possible, I have made suggestions on how to get the best sound for the least money. Following on from the principle that it's easier and more effective to make a high-quality recording than to fix a low-quality recording, I recommend spending more money on the recording stage of your podcast than the editing stage. As such, I advise you in Chapter 2, "Gear," to invest in some good-quality recording equipment, but then I demonstrate editing techniques using the very basic audio editing program *Audacity*. Audacity might not be the best audio editing program, but it's the most accessible – it is free, it runs on Windows and Mac, and it is adequate for creating a basic podcast. If you wish to undertake more advanced editing, then I discuss a few other options in Chapter 8, "Editing."

Throughout the book, I demonstrate techniques using specific pieces of audio equipment and computer programs, but you might already have a setup. The audio techniques and principles that I am discussing apply to similar equipment and programs.

Some people might find the idea of presenting a very polished podcast as being inauthentic. They might feel that removing some of the rough edges compromises their DIY ethic. Why might people feel this way? It's because technical choices are a part of the creative process. While everyone needs to know the basics, technical choices can be an important part of the storytelling, helping you create atmosphere and individual character. I don't want you to see this book as a series of rules about recording and editing. Instead, I hope to demonstrate that a good understanding of the technical side of podcasting will enable you to express yourself and to reach the people who are important to you. I have included audio clips throughout so that you can hear the outcome of technical choices and learn to be the judge of your own work. For those who are reading the print version of this book, the audio files are located at `https://www.transducer-audio.com/audio-files-for-book`. Saying that, there is a focus on creating clear recordings of human speech, so as to make podcasts accessible. Not every podcast has to sound like it was produced in a professional studio, but every podcaster should make the utmost effort to record and edit speech so that it's intelligible to the listener. Podcasting has become an important way for people to share information, and I would like to see as many people included in that as possible. I will be discussing issues of accessibility throughout the book. Once you have made a high-quality recording, you can present it in any way that you like.

Everyone has to begin somewhere. Some of you will be new to podcasting and some more established. No matter where you are in your journey, I would like you to join me in the next chapter at a time when I was just starting out in radio so that I can share an important lesson that I learned.

CHAPTER 1

File Formats and Settings

A good understanding of file formats and settings can make a huge difference to your audio quality. When working with clients, this is the number one issue that I encounter, so I've put it at the front of the book. No matter what equipment you're using, this is the easiest way to improve the audio quality of your podcast.

I learned the importance of understanding file formats and settings the hard way, while making a radio documentary early in my career.

I had met a woman called Aunty Dawn Daylight. Aunty Dawn Daylight is Aboriginal; she is a proud Jagera-Turrbal woman and a respected elder who is a fixture of Brisbane's West End Community. She looks out for others and is always ready with a joke or a song.

Dawn's childhood experience was shocking – she'd been held as a child slave at my former Catholic high school. Some may find the term "slave" too strong, but Dawn had been taken away from her family, imprisoned, and forced to work without pay. What's worse is that this was a perfectly legal thing to do to an Aboriginal child in the state of Queensland in Australia at the time.

Dawn had a lot of unanswered questions, and I was in a unique position to find some answers because one of the nuns who looked after the school's archival collection was a family friend. I got in contact with the school's archival organization, and they agreed to speak with me

C. M. Green, *The Podcaster's Audio Handbook*,
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and to show me around the convent. Previously, I had been all over the school, but I had never before entered this area. I can still recall my visceral reaction to finding child-sized cells. I suppose to the nuns they looked like ordinary dormitories, just with higher security. I imagined how they would look to a child who was staring out through the bars wondering why she'd been taken away from her family.

I really wanted to do right by Dawn. I spent a year researching and refining the story. I hoped that the radio documentary could raise awareness for the **Stolen Wages** campaign, which aims to restore the wages of the many Aboriginal people in Queensland who had been forced to work without pay.

Despite all my work, the harsh truth is that the documentary is almost unlistenable. In some places, you can't even make out the words. I released it with an accompanying transcript so that people could follow along.

One of the main reasons the recordings were so bad was because I didn't have a good understanding of file formats and settings. I recorded all of the interviews in the mp3 format. I can't even go back and repair the recordings because I didn't record enough audio data.

Thankfully somebody else made a follow-up documentary with Aunty Dawn. What follows is a technical guide so that you can give full justice to the stories that matter to you.

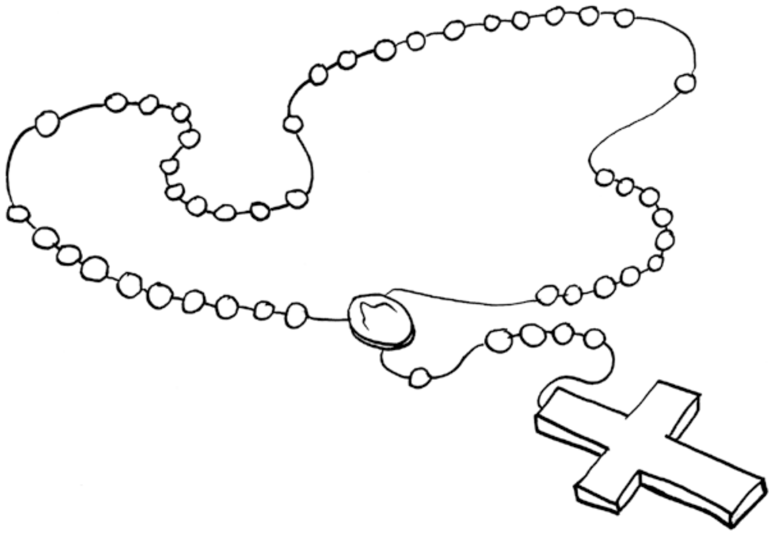

Figure 1-1. *Aunty Dawn described the nuns: "You used to always hear them coming with their rosary beads and keys"*

A good understanding of file formats and settings will help you create files that strike the right balance between size and quality.

In this chapter, I explain some basic audio concepts. I discuss compressed and uncompressed formats, bit rate, sample rate, and stereo vs. mono. These are all concepts that you will encounter in the recording and editing of a podcast. The concepts I am discussing in this chapter apply to recording using a computer or a portable recorder. I go into the settings on a portable recorder in more detail in Chapter 6, "Recording Outside" in the subsection "Settings on a Portable Recorder".

There are two types of learners: people who need to know the reason for things and people who just want to get to the point. I have prepared a summary of this chapter for the latter type of learner where I simply list the recommended file formats and settings. You can find it in this Chapter in the subsection "Just Give Me the Executive Summary". If you're like me and you need to understand something to be able to use it, then read on.

Figure 1-2. *Me making breakfast*

What Is Audio?

When you're working with audio, it pays to know the basics.

Sound moves in waves, like the ripples caused when you drop a rock in a pool of water.

Generally speaking, audio is sound that has been translated into electricity. From there, it can be stored in a number of ways. For example, it can be physically etched into the grooves of a vinyl record. Audio can be reproduced, manipulated, and transmitted. At the end of the process, the audio is played back through a speaker and once again becomes sound.

The squiggly line that you see in your editing program represents the audio waveform (see Figure 1-3).

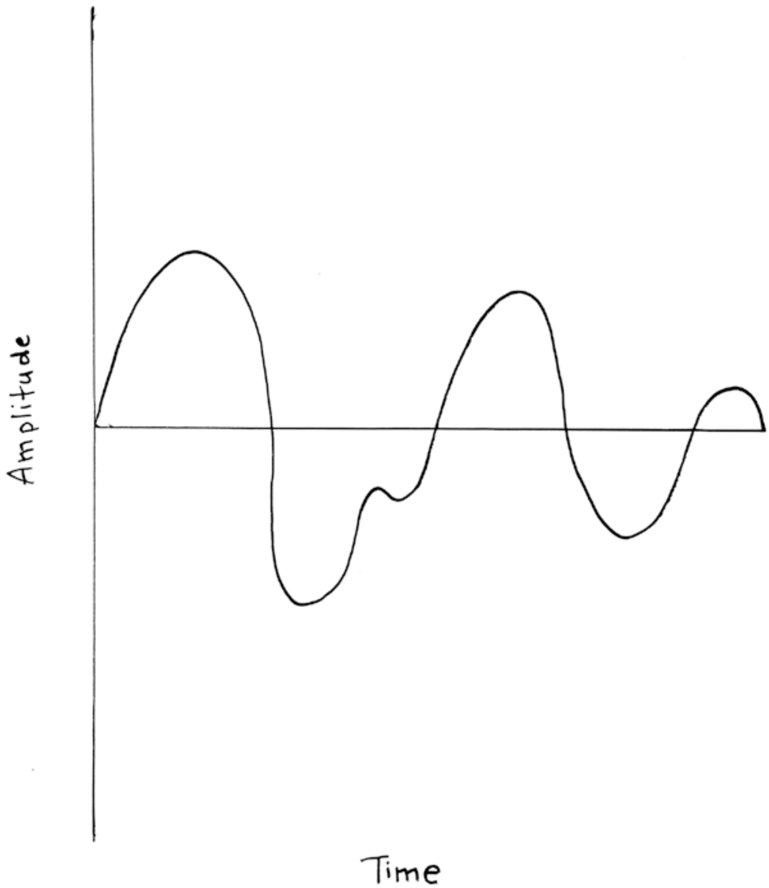

Figure 1-3. *This is a grossly simplified representation of an audio waveform*

This representation of an audio waveform has been magnified many times. If it were a real audio clip, it would be very short.

If you've ever worked with any kind of audio, you probably already know that the horizontal line in the middle is a time value that moves from left to right.

The vertical line that I have marked "amplitude" is the strength of the signal. A strong signal, which looks like a big wave, more or less means a loud signal. I'll go into this in more detail later.

The audio waveform moves back and forth above and below the horizontal line. You can think of this as your speaker cone (Figure 1-4) moving in and out, pushing the air in and out, and causing a vibration. The round object that you can see on the front of your speaker is a speaker cone.

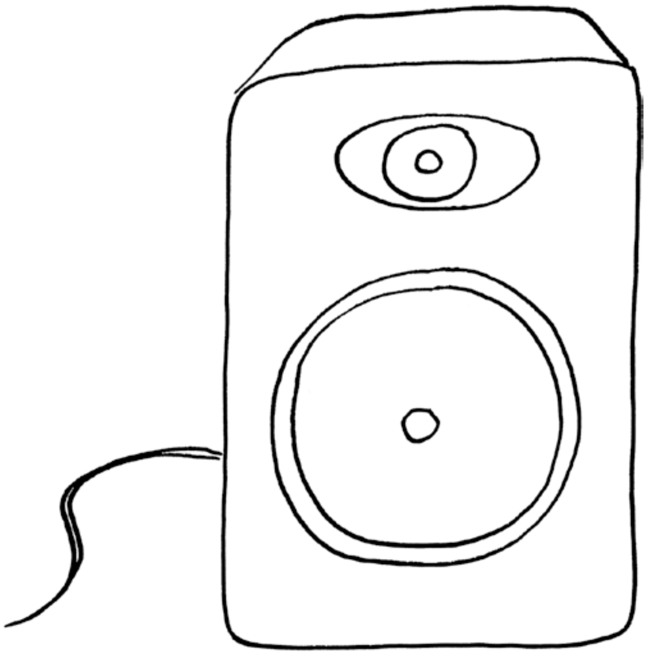

Figure 1-4. *Speaker cones in their natural habitat*

When the waveform is far above the horizontal line, your speaker cone is pushed far out. When the waveform is far below the horizontal line, your speaker cone is pushed far in. At the point that the waveform hits the middle line, the speaker cone is in the middle. If the idea of the speaker cone causing waves by physically pushing air is hard to visualize, then imagine that it's pushing water as in Figure 1-5.

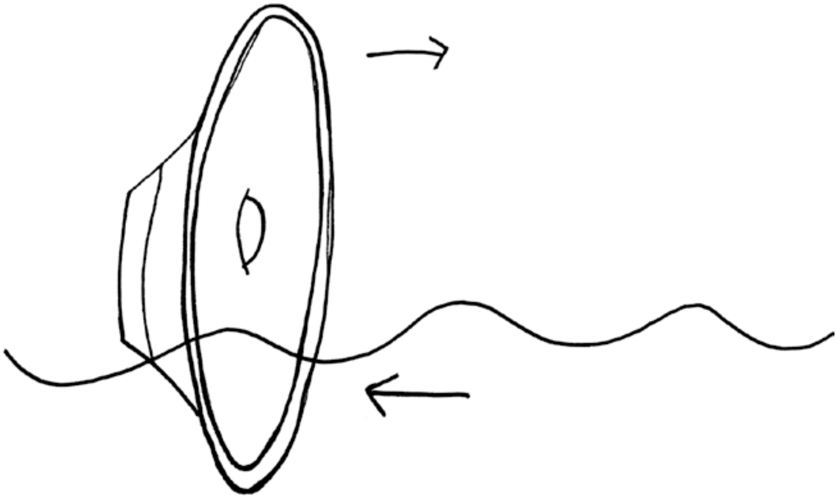

Figure 1-5. *A speaker cone creating a wave by moving in and out*

This all happens very fast, at the speed of a vibration. If you have a speaker at your house, you can put on your favorite tune, crank the bass, and see the speaker cones vibrating.

So, audio is sound that has been translated into electricity, from which point it can be stored in a number of ways, reproduced, manipulated, transmitted, and played back.

Audio can be analog or digital. Vinyl records are one example of an analog audio technology that is still in use today. The difference between analog and digital is that analog waveforms are continuous, whereas digital waveforms are not.

Digital Audio, the Sample Rate, and the Bit Rate

The audio on a CD or a computer is digital audio. The audio on your computer starts off as analog and is converted to digital by an analog-to-digital converter. Digital audio is then once again converted back into analog for playback on your speakers.

Digital audio is different to analog audio in that rather than recording a continuous waveform, the computer samples the waveform. When you set your sample rate to 44,100 Hz (or 44.1 kHz), you are telling the analog-to-digital converter to take a snapshot or a "sample" of the analog audio 44,100 times every second. The data is not stored as a continuous waveform as it would be if it were analog. Instead, it is stored as a series of numbers or points of data.

If you zoom really closely into an audio file in Audacity, you can see the samples (see Figure 1-6).

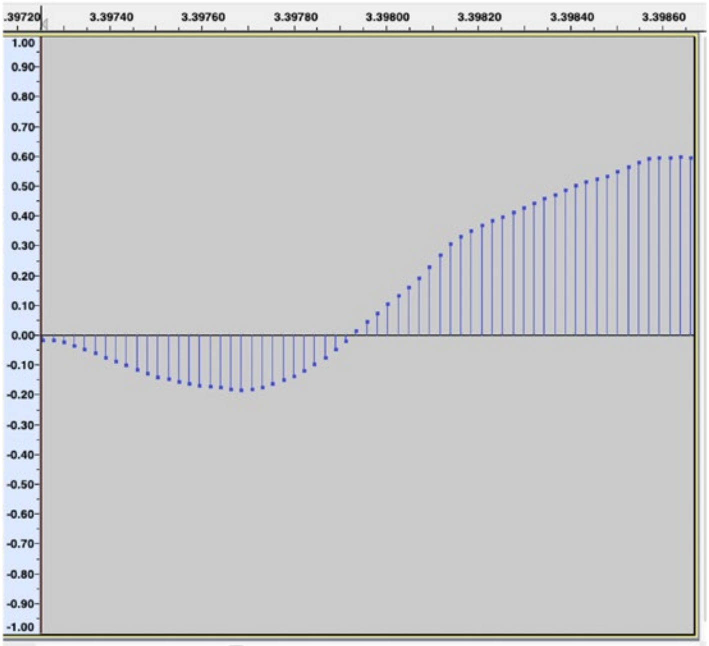

Figure 1-6. *An extreme close-up of an audio file in Audacity*

There are two settings that will determine the detail with which your computer will convert analog audio into digital audio: the sample rate and the bit rate. The sample rate is connected to the time measure or the horizontal axis. You can remember that because it's a certain number of samples per second. In Figure 1-6, it is represented by the series of evenly spaced vertical lines. The bit rate determines the detail with which the computer records each sample.

If you put the sample rate and the bit rate together, you have a highly detailed grid on which a computer records points of data. I have created a simplified version of this in Figure 1-7.

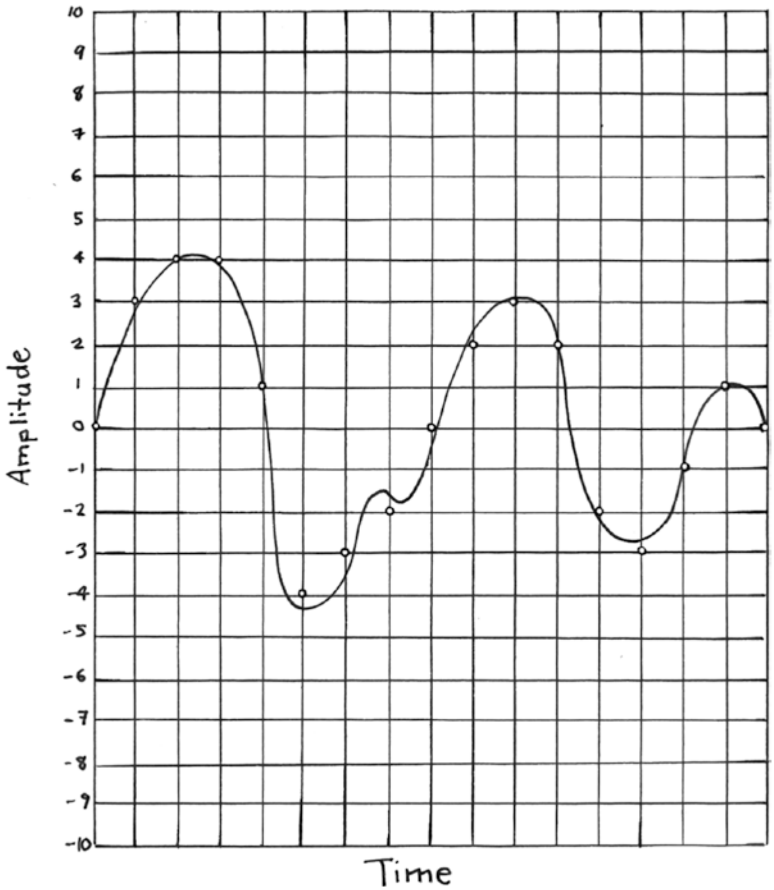

Figure 1-7. *This is a grossly simplified representation of the conversion of analog audio into digital audio*

In Figure 1-7, the analog audio is represented by the continuous line. The digital audio is represented by the series of dots. The sample rate is represented by the vertical lines of the grid, and the bit rate is represented by the horizontal lines.

I have "sampled" the analog audio by assigning a series of dots to it. For every sample there is one dot, which has been placed on the nearest horizontal line. I would record these dots as a series of numbers. The data for Figure 1-7 would look like 0, 3, 4, 4, 1, -4, etc. On a computer, the numbers would be much larger and would be stored in binary code.

You'll notice that the dots on Figure 1-7 don't always match up with the continuous line. This is what is happening when you convert analog audio to digital: you're only getting an approximate picture. When the computer converts digital audio back into analog to play on your speakers, it fills in the spaces as best it can.

When you're setting the sample rate and the bit rate, you're setting how detailed the grid will be and so how closely the series of samples will match the analog audio signal. Higher values will give you a more accurate conversion, but will require more computer memory.

As podcasts are streamed over the Internet, it is necessary to find a balance between file size and audio quality. The stream of a small file will load quickly, whereas the stream of a large file will take much longer to load. A good sample rate to use for the entire podcasting process is 44.1 kHz. And 24 bits is a good amount of detail for recording and editing. After you have finished editing, you will need to *bounce* your files, which is a process that combines all the files in your session into a single file. This final file is known as the *mixdown*. Bounce your files to a bit rate of 16 bits for listening back.

This simplified explanation of digital audio will help you find a balance between file size and audio quality, but there are more factors to consider.

Compressed and Uncompressed File Formats: wav vs. mp3

When working with digital audio, it is important to understand the difference between compressed and uncompressed file formats so you can know which to use when. In this context, an audio file that has been compressed[1] has had a whole lot of data taken out of it to make it smaller. An uncompressed audio file is in its original form. A wav file is uncompressed audio, and an mp3 is highly compressed.

Think of the wav file as a picture of a happy whale, as depicted in Figure 1-8.

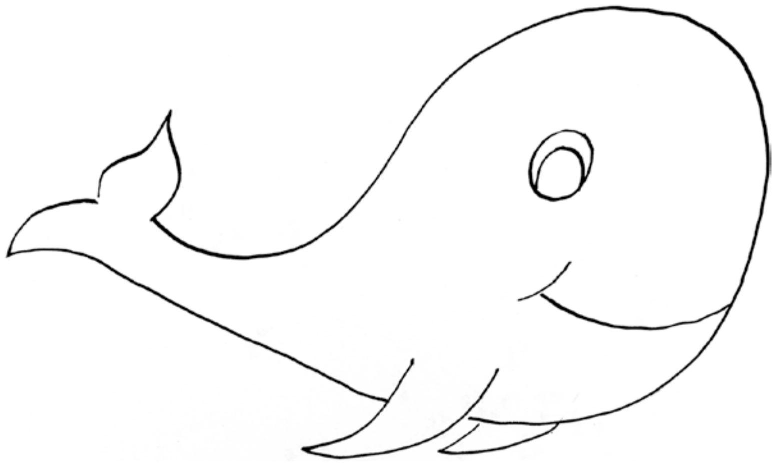

Figure 1-8. *A picture of a happy whale*

We already know from the previous section that the wav is made of lots of tiny little dots or points of data. There are so many that they look like a continuous line.

[1] "Compressed" also describes another important audio concept.

Now think of the mp3 as the same whale in the form of a connect-the-dots puzzle as in Figure 1-9. It is missing some data.

Figure 1-9. *The happy whale picture has been converted into a connect-the-dots puzzle*

An mp3 is about 10% of the size of a wav. This makes it useful for sending over the Internet, but the compromise is in the audio quality. Saying that, it is truly remarkable that it's possible to remove this much audio data and still have a listenable file. How does this work? The answer is psychoacoustics, or the science of how your brain interprets sound.

To explain psychoacoustics, I want to go back to the connect-the-dots puzzle. A person who didn't know anything about whales or connect-the-dots puzzles might fill in the puzzle and come up with something like in Figure 1-10.

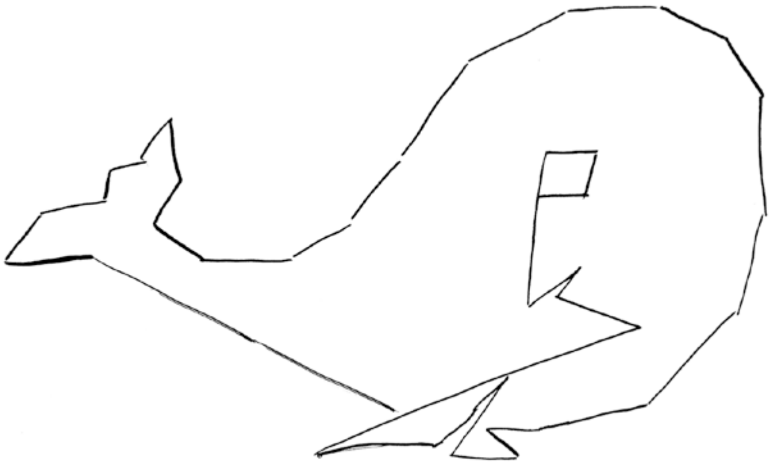

Figure 1-10. *It's not quite right*

Most people, however, would be able to take that puzzle and turn it into a pretty good whale based on prior knowledge. The mp3 works in a similar way. It gives you just enough data and allows your brain to fill in the rest.

So a wav file is an uncompressed audio file, and an mp3 is highly compressed. The wav and the mp3 formats are both useful, but in different parts of the podcasting process.

Working with Files in the wav and mp3 Formats

You should work with the uncompressed wav format for almost the entirety of the podcasting process. At the end, you should make an extra copy of your work in the compressed mp3 format to send over the Internet. This will enable you to maximize the amount of audio data that you're working with and distributing to your listeners.

Just to be clear, you should

- Record in wav
- Edit in wav
- Save your final work as a wav
- Export a copy of your final work as an mp3 file

The reason that you should record and edit in the wav format is that these parts of the process require large, detailed files. You should create a final copy of your podcast as a wav because you never know when you're going to need it. You might get an offer to play your work on radio. Or you might want to come back years later and use a section of your final show. I can foresee a time when the Internet will be fast enough for people to upload podcasts in the wav format. It's good to have a high-quality backup of all your hard work.

The reason that you should create an mp3 version of your podcast is that the smaller file size of the mp3 makes it easier to send the file over the Internet.

When making your mp3, you should once again consider the balance between audio quality and file size. Setting the mp3 to 128 kbps strikes a good balance.

When making an mp3, choose "constant bit rate" over "variable bit rate," because this will be compatible with more playback systems.

It's impressive that mp3s can sound reasonably good with only 10% of the data of a wav. However, creating an mp3 is a trick that you should only perform once at the end of your recording and editing process. Repeatedly saving a file in the mp3 format is like taking a photocopy of a photocopy: the audio quality is quickly going to degrade. If you've received a sound file that's in the mp3 format, the best thing you can do is save it as a wav as soon as possible to reduce any further loss of quality.

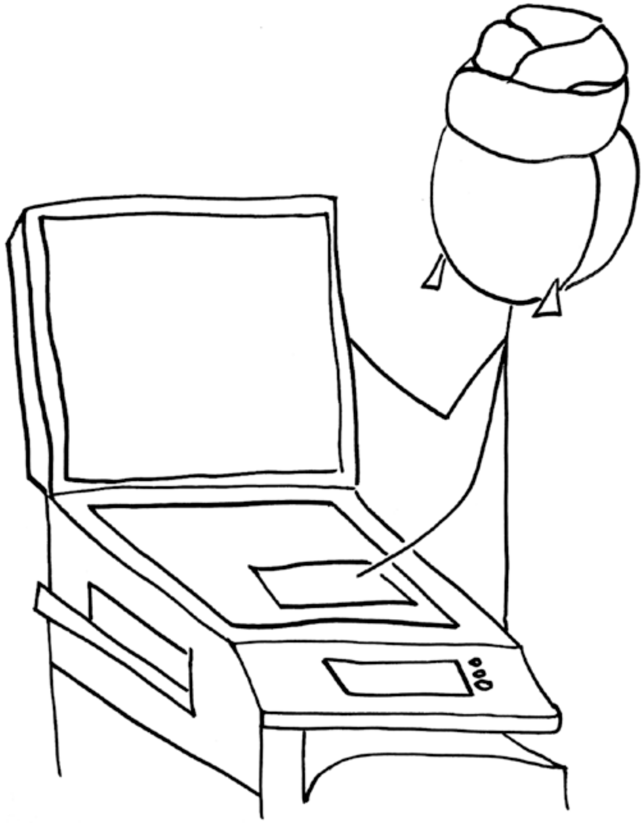

Figure 1-11. *Repeatedly saving a file in the mp3 format is like taking a photocopy of a photocopy*

To summarize, wav is uncompressed audio, mp3 is compressed. Record in wav, edit in wav, save the final in wav, and export as an mp3 for transfer of the file over the Internet.

Stereo vs. Mono

Podcasters may wonder whether using a stereo or a mono file is a good choice for their work. Again, it depends on your context. Stereo is useful in certain circumstances, but it increases your file size and is more complicated to work with.

A stereo file has two channels: left and right. It can create a sense of movement, or it can make you feel like you're enveloped in a physical space. A mono file has only one channel: the center.

Stereo works well for a sound such as a passing vehicle. If you listen to the audio clip in Audio 1-1 while wearing headphones, you can appreciate the effect.

Audio 1-1 Using stereo to create a sense of movement

Stereo can give you a sense of a physical space. I took my portable recorder to a road underpass and recorded the sound of the cars passing overhead. I have provided the file in both stereo (Audio 1-2) and mono (Audio 1-3). In the stereo version, you can hear the sound bouncing off all the concrete surfaces to create the image of the physical space. The mono version is missing that. The effect works best when you listen to the audio clip using headphones.

Audio 1-2 Using stereo to create a sense of physical space

Audio 1-3 The same file in mono lacks a sense of physical space

The sense of physical space that you get with stereo files works well for music. You can have the singer in the center, the guitarist to the left, the bassist to the right, and the drummer up the back.

Stereo files are cool, but mono is a seriously underrated format. Mono files only require one channel of audio, so they are smaller and easier to work with. You might think that nobody listens to mono recordings anymore, but you would be wrong. Any recording that comes out of a single speaker – such as the speaker of many models of smartphone – is mono. Another reason to favor mono recordings is accessibility. Erin Kyan from **The Love and Luck Podcast** mixes his podcast in mono to accommodate people who can only hear from one ear.[2]

[2] Audiocraft, 2018. *Under the Hood with Love and Luck*. [podcast] Audiocraft Podcast. Available at: `https://play.acast.com/s/audiocraft/underthehoodwithloveandluck`

If you're recording a person talking, record this on a single channel, or in mono. If you use the stereo format to record a single person talking, then most times you're going to pan it to the center, which doesn't take advantage of what stereo can offer. I've provided an example of this in Figure 1-12.

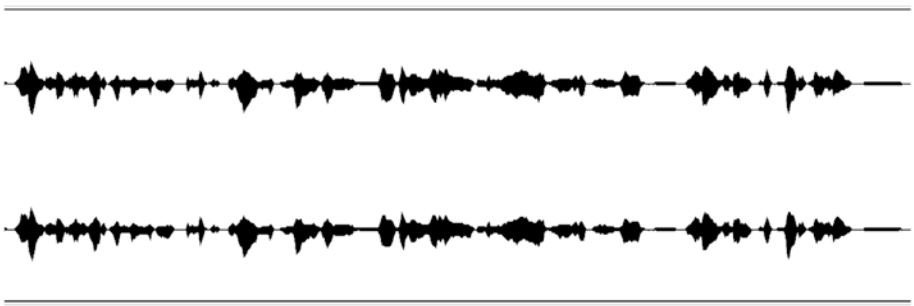

Figure 1-12. *This audio file of a single person talking has been recorded in stereo*

You can see from the waveform of Figure 1-12 that the two channels (left and right) are identical. As there's no difference between the two channels, there is no stereo effect. In this case, I would recommend that you split the file into mono and only use one side of it. You might need to pan the new file to the center in your audio editing program.

Recording one person talking on one channel makes sense, but what about if you're recording two people talking? Usually you will be panning both people to the center, so you won't need to use a stereo file. If you are recording two people talking, then record them with two microphones on two separate mono channels.

There's an exception to this. If you are recording using the inbuilt microphones of a portable recorder, you will have to record in stereo, no matter how many people are talking. Portable recorders require you to record these files in stereo so that they can filter out the sound of their own operation.

After you have finished editing your podcast, you will need to *bounce* your files, which is a process that combines all the files in your session into a single file. Consider whether you will bounce your final mixdown to stereo or mono. If your podcast is heavy on music and sound effects, use stereo. If your podcast is mostly talking, then mono is a better option.

When creating a podcast, you should choose the stereo format if you want to create a sense of physical space or movement. Otherwise you should consider mono files as they are smaller, more accessible, and easier to work with.

Those of us who need to know the reason for things have now caught up with the other group, the people who don't need to know how a toaster works to make a piece of toast. They've already had breakfast, and they're ready to make a podcast. Here's the summary I promised. I am sure that you will find it useful as well.

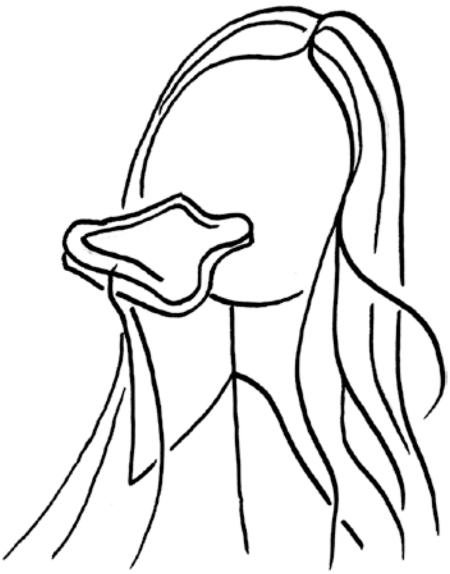

Figure 1-13. *The sort of people who don't need to know how a toaster works to make a piece of toast*

Just Give Me the Executive Summary

We have so far discussed a number of file formats and settings – wav, mp3, bit rate, sample rate, stereo, and mono – but how should you apply them to your podcast? I've created a summary that you can refer to in your recording and editing process.

If you're recording people talking, choose the following settings:

- Wav
- Bit rate of 24 bits
- Sample rate of 44.1 kHz
- Mono

Set up your editing session in your audio editing program as follows:

- Wav (this might be default)
- Bit rate of 24 bits
- Sample rate of 44.1 kHz

After you have finished editing, you will need to *bounce* your files, which is a process that combines all the files in your session into a single file. This final file is known as the mixdown.

Bounce your final mixdown to

- Wav
- Bit rate of 16 bits
- Sample rate of 44.1 kHz
- Mono for a podcast that's mostly talking
- Stereo for a podcast with music and sound effects

A copy of the final mixdown should be exported as

- mp3
- 128 kbps
- Constant bit rate
- Sample rate of 44.1 kHz
- Mono for a podcast that's mostly talking
- Stereo for a podcast with music and sound effects

Choosing File Formats and Settings in Audacity

The audio techniques and principles I am discussing in this book apply to most audio equipment and audio editing programs, but as previously mentioned, I will be demonstrating them in Audacity.

Setting the Bit Rate and Sample Rate in Audacity

You can set your default sample rate and bit rate in Audacity by going into the Preferences menu. To get there on a Mac, go to the Audacity menu and choose Preferences. On a PC, go to the Edit menu and choose Preferences. A window will open as in Figure 1-14. Choose the submenu labeled "Quality" on the left.

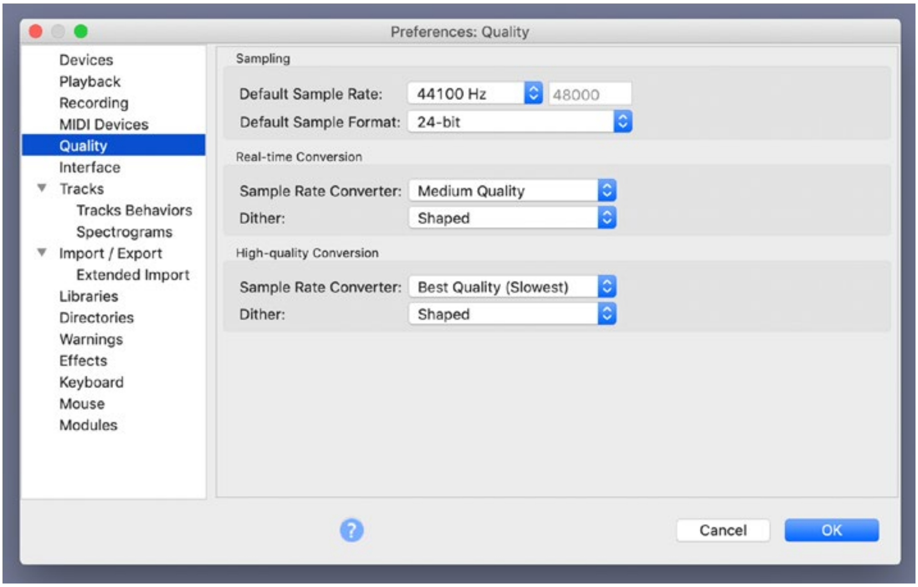

Figure 1-14. *The Audacity Preferences menu*

Audacity refers to the bit rate as the "Default Sample Format."

Many people would think that after you had set your preferences, Audacity would respect them. Those people would be wrong. When setting up an editing session in Audacity, you should note that even though you may have set your sample rate preference for a default session, Audacity will change the sample rate of any session to that of the first audio file that you import. So if you are making a recording, Audacity will respect your wishes as to the sample rate, but if you are importing a file, it will assume that it knows what you want more than you do. In this respect, Audacity is acting a bit like the autocorrect function on your phone. I find the whole thing ducking irritating. After importing your file, you will find Audacity's new project sample rate in the drop-down menu in the bottom left-hand corner as depicted in Figure 1-15.

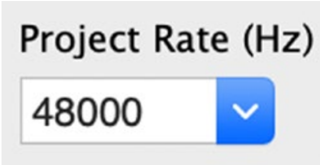

Figure 1-15. *The drop-down menu showing a project's sample rate in Audacity*

You can change the sample rate of your project back to the one that you desire using that same drop-down menu.

Thankfully, Audacity doesn't also change the bit rate of your project.

Choosing Stereo or Mono in Audacity

You can record in Audacity in stereo or mono. Go to the Tracks menu and select "Add New," and then choose mono or stereo as required. See Figure 1-16.

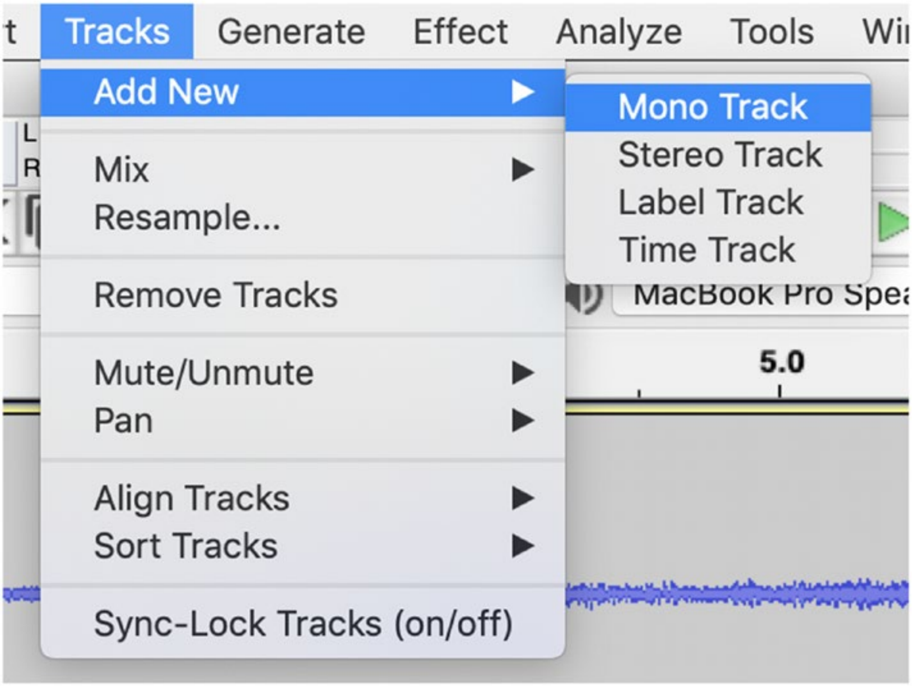

Figure 1-16. *Creating a mono or stereo track in Audacity*

Audio files that are imported into Audacity will already be in stereo or mono.

If you made a recording of a person talking in stereo and now want to convert that file into mono, you will need to split the file. This option can be found in a menu to the left of your track. Click the name of the track, and the menu will appear. Choose “Split Stereo to Mono” as in Figure 1-17. Your single stereo file should now be two mono files. Delete one of the files.

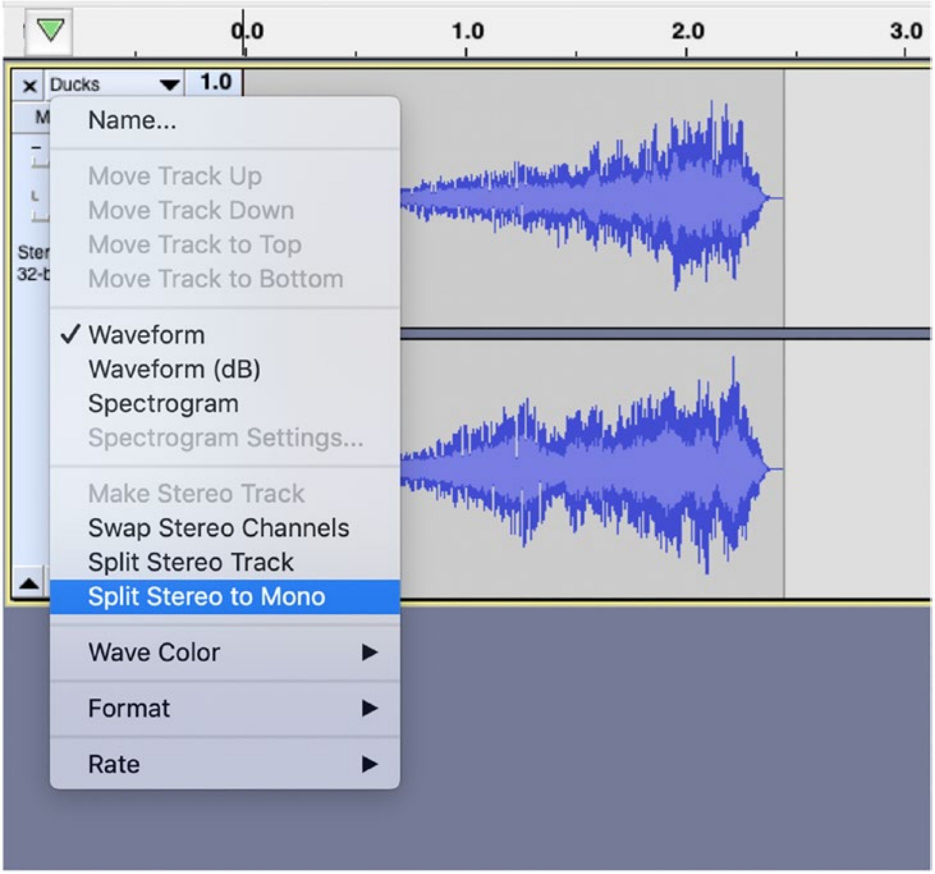

Figure 1-17. *The function to split a stereo track into two mono tracks*

Bouncing Your Files in Audacity

I will go into editing in much more detail in Chapter 8, "Editing." For now, you should know that when you have finished editing your podcast, you will need to bounce your files. Audacity refers to this as exporting.

To bounce your files in Audacity, go to File ➤ Export ➤ Export as WAV as in Figure 1-18.

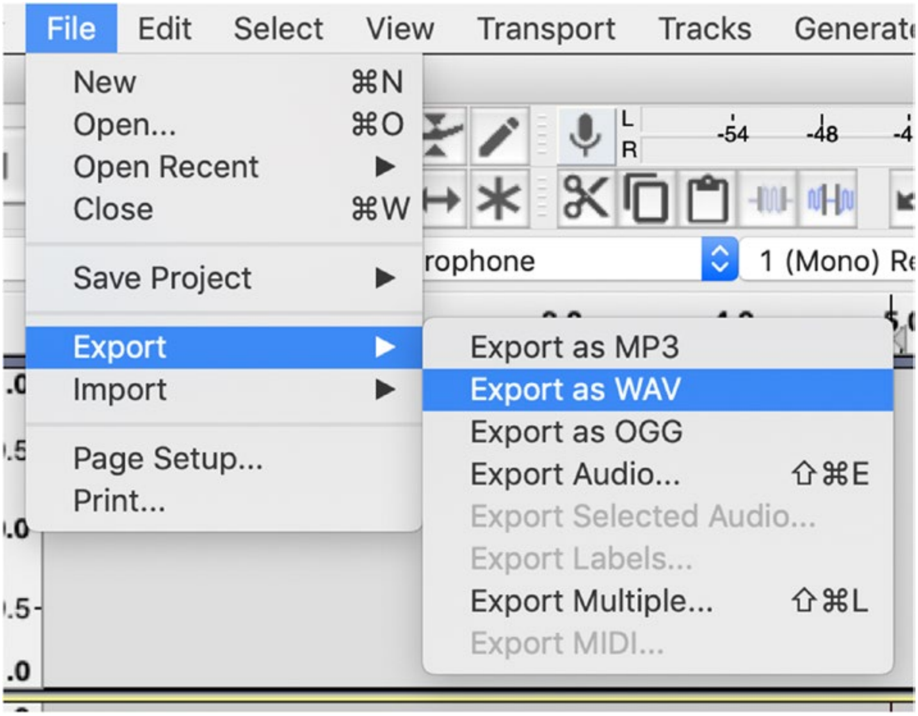

Figure 1-18. *Export as wav*

A dialog box will come up which will enable you to choose where to save your files and to choose the bit rate at which to save them. There are a number of options available, and "WAV (Microsoft) signed 16-bit PCM" (as depicted in Figure 1-19) is the correct one.

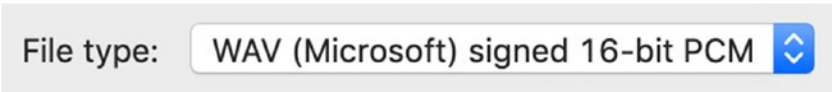

Figure 1-19. *Choose your bit rate*

You will also need to bounce the final file as an mp3 to send over the Internet. The dialog box for an mp3 has many options as shown in Figure 1-20. This is where you set the file to a constant bit rate, choose the quality, and decide between stereo and mono. If you are making a stereo file, choose "Joint Stereo."

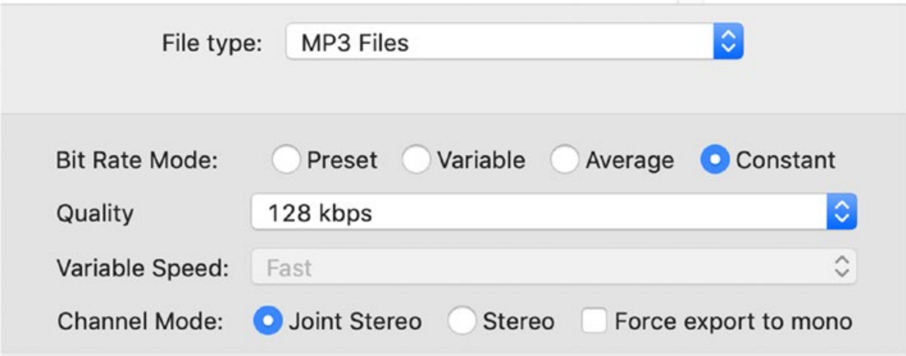

Figure 1-20. *Options available for exporting your files as an mp3 in Audacity*

Choosing the sample rate and the bit rate for your project and exporting your files should be similar in most audio editing programs. You can also choose many of these options on other audio equipment such as a portable recorder. I will discuss how to set up a portable recorder in Chapter 6, "Recording Outside."

No matter what equipment you're using, the easiest way to optimize your audio quality is by understanding file formats and settings and how they apply to your podcast.

CHAPTER 2

Gear Part 1

One of the first questions new clients ask when they approach me to produce their podcast is, "What would be a good setup for a podcaster?" A lot of the time the discussion reveals that the real question is: "How can I make a podcast sound good without spending too much money on audio equipment?" I get it: you're on a budget; we're all on a budget; Oprah brings a packed lunch to work. One of the great things about podcasting is that it's accessible to a wide range of people, as long as they have an Internet connection. The barriers to entry are low, but to create a podcast with good audio quality, there must be some investment in gear.

C. M. Green, *The Podcaster's Audio Handbook*,
https://doi.org/10.1007/978-1-4842-7361-6_2

Figure 2-1. *You'll probably have to spend some money to create a podcast*

This is the first of two chapters about gear. I've placed these chapters early, because I continually refer to this audio equipment throughout the book. Even if you don't intend on buying any gear, these chapters are worth reading because they can help you understand the equipment that you're using. The chapters are quite in-depth, so you might want to skim them to get an overview and come back to read the details as you need them.

Using appropriate audio equipment will make the process of recording and editing your podcast a lot simpler than if you use equipment that is not well suited to podcasting – and the final product will be of a much higher standard. Using the right gear will definitely help you make a good recording, but no matter what gear you're using the information contained within this chapter and this book should enable you to improve the audio quality of your podcast.

My recommendations for gear consider cost, audio quality, and the challenges presented by recording at home. The equipment I'm recommending isn't exactly cheap, but in my opinion, it's the least money that you can get away with spending if you want to make a polished recording. There's a great variety of cheaper equipment available – much of it aimed toward podcasters – but I don't presume that you're reading this guide to learn how to make a bad recording. If you have a small budget, then consider hiring a studio or hiring equipment. These options should work out better in terms of quality and cost.

I have explained the reasons for my recommendations to enable you to make informed choices about your equipment purchases. Your choices should take into account the environment in which you'll be recording and the overall sound that you want to achieve.

The biggest challenge for recording at home is managing background noise, so I have focused on equipment that minimizes this problem. This will ultimately save you money as you don't have to invest in expensive sound proofing treatments. It will also enable you to make a decent recording in the field.

In this chapter, I will look at recording using smartphones, an audio interface, and a portable recorder. I will also look at headphones and cables. I cover microphones and their accessories in their own chapter because a microphone is a key piece of equipment and it is useful to have a thorough understanding of how it works.

I've included audio clips so that you can hear the quality of the gear in action. I haven't added any *processing* to the clips except to boost the gain and to fade all of the clips in and out. Processing is a general term that describes altering the characteristic of an audio signal. Examples of processing are compression and equalization.

I have based recommendations on my industry experience, and I have also tested out a selection of the latest gear to make sure that I am bringing you the most up-to-date information in a rapidly changing field.

Before I get started on equipment recommendations, I want to talk about background noise.

Dealing with Background Noise When Recording at Home

Some microphones, including most of the cheaper microphones, pick up a lot of background noise. People who are using sensitive microphones to record at home face the maddening task of trying to sound proof a noisy environment. Before I get started on equipment recommendations, I want to explain why I think that it's much easier, cheaper, and more effective to make a good recording at home using equipment that is designed to reject background noise.

Background noise is a particular problem for podcasters who are recording at home because talking isn't very loud. If you are recording your band and you place a microphone in front of an amp that's blaring out electric guitar, that sound will probably overwhelm most background noise. However, there's much less difference between the loudness of the background noise you would typically hear at a home and the loudness of speech. Podcasters will know that if you are speaking into a microphone and someone nearby starts a lawn mower, there's a good chance you will record the lawn mower as well, especially if you are using a very sensitive microphone.

There are certain common sense actions that you can take to reduce the amount of background noise in a room, such as shutting the windows. I'll go into this in more detail in the next chapter. However, re-creating the level of sound isolation that you would find in a studio in your own home is a difficult and expensive undertaking. There's a good reason that studios are quiet. One strategy that studios might use to create sound isolation is to build a room within a room. In this design, the inner room is physically isolated from the outer room as much as possible. Even the floor floats on rubber pads or springs. I have created a rough picture of this concept in Figure 2-2.

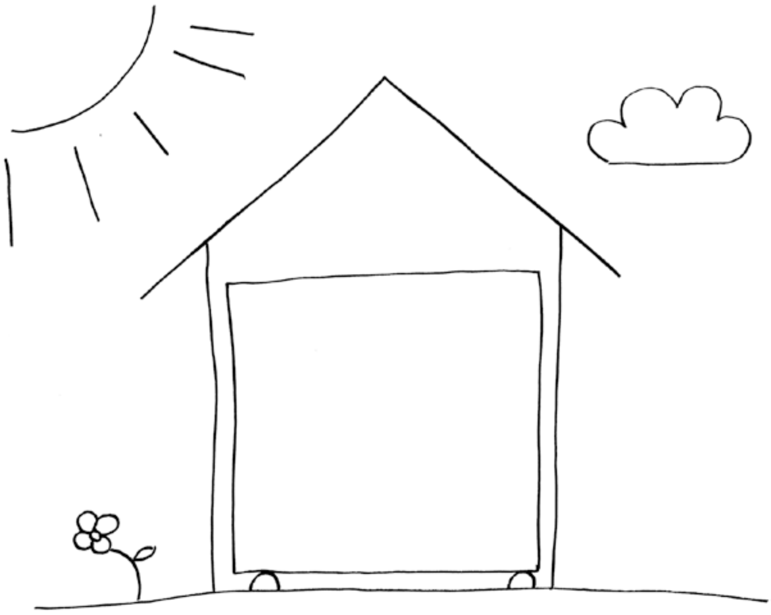

Figure 2-2. *Creating a room within a room is a strategy that recording studios employ to improve sound isolation*

There is a spectrum of actions that you can undertake to improve sound proofing in your home that sit between shutting the windows and creating a room within a room. For example, I've seen people record in rooms where they've hung a blanket on the door because they've found that it reduces background noise. This is fine, but it's a slippery slope. If you are trying to use a very sensitive microphone to record speech at home, you could quickly end up creating a space that's solely dedicated to recording. I don't know your particular situation, but you should consider whether it's practical to even set aside a room for this purpose. You'll have to consider how much you're willing to invest: is podcasting a hobby or your career?

There are a few ways to tackle the issue of background noise when recording at home that do not involve building a studio. You could

a) Accept that there will be background noise in your podcast because you're recording at home and the background noise simply adds to the vibe of your podcast

b) Hire a studio because you want the perfect recording, and ultimately hiring a studio will be cheaper than creating one in your own home

c) Invest in equipment that rejects background noise and make some effort to sound proof your home

I will be focusing on option C. If you start off using equipment that is designed to reject background noise, you can create a fairly high-quality recording in your home without having to invest too much money or effort in sound proofing. But before I get to that, I want to discuss the popular idea that you can create a quality podcast using your smartphone.

When to Record on a Smartphone

If you already own a smartphone, then this will be the cheapest way to record your podcast, but it will sound dreadful. When you're recording with a smartphone, you'll be fighting background noise, particularly if you're recording in the field. It will be near impossible to fix the sound quality in post production, because you won't have recorded enough quality audio data. A lot of post production is about taking the bad audio out, but that relies on having a sufficient amount of good audio present. Your smartphone will not provide this.

The only reason that I'm bringing up smartphones is because there is one situation in which I would use one to make a recording: if I came across some incredible content and there was no other option. Like if I was out for a walk and I ran into Beyoncé having a rap battle with the Queen. If the content is compelling enough, then the audience will be forgiving of poor audio quality and will give their full attention to trying to make out what you have recorded. Otherwise, save your smartphone for its intended purpose: posting selfies on social media.

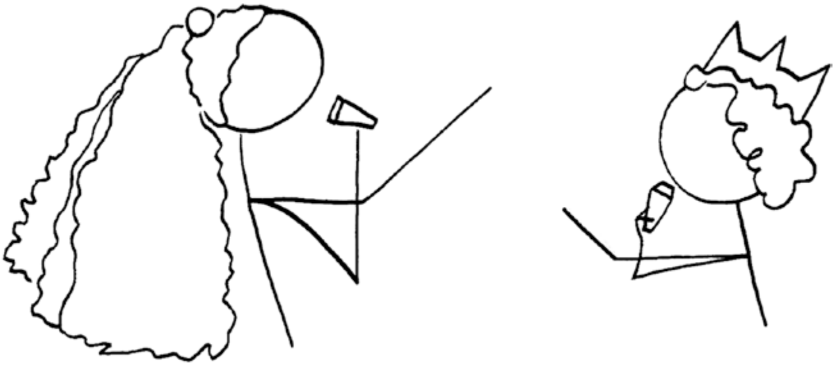

Figure 2-3. *Only record audio for a podcast on your smartphone if you come across amazing content and have no other options*

Audio Interface: The Analog to Digital Connection

If you want to record high-quality audio and you only ever record at home, then you should invest in an audio interface. An audio interface is a piece of equipment that takes an analog signal from your microphone, makes the signal stronger, and converts it into a digital format for your computer. It also does this in reverse, converting digital audio from your computer back into analog for playback with headphones or speakers. There are microphones available that will plug into your computer without a dedicated audio interface, but using these presents a number of compromises. I will discuss this option in the following chapter.

I'm often asked why you can't just use an adaptor to plug a regular microphone into a computer. Fair question. There are two reasons: the signal is too weak, and the microphone signal is analog, whereas the computer only understands digital formats.

An audio interface has components called *preamps* which take the weak microphone signal and make it stronger. Think of running audio through a preamp as like giving it a healthy meal. A stronger signal more or less translates into a louder signal. If you feed a weak signal into your computer, then it is possible to turn it up digitally, but this would significantly compromise quality. That's more like feeding your audio on junk food. The quality of the preamps is a key distinction between a decent piece of audio equipment and a more affordable one.

The other function of an audio interface is to turn the analog signal of the microphone into a digital signal. This is done with an analog-to-digital converter. The quality of this component is important as well. You want your analog-to-digital converters to be both fast and accurate. Interfaces can come with other features, but the preamps and the analog-to-digital converters make up the key functionality.

You can buy mixing desks that will function as an interface and allow you to mix together and process different signals on the fly. The mixing desk might have signal processing functions such as equalization and compression which are great if you know how to use them skillfully. These functions are also available in your audio editing software. A mixing desk is useful for a live show, but it is overkill for your typical podcast. As most podcasts are prerecorded, I recommend that you simply use an interface to get a quality recording of one or more microphones and then later do all of your mixing and signal processing using an audio editing program. You can still do a "live" style show with multiple people using an interface, because you can buy interfaces that will accommodate a large number of microphones. As long as you are not actually livestreaming, you can record your “live” show on to your computer and edit it as much or as little as you like.

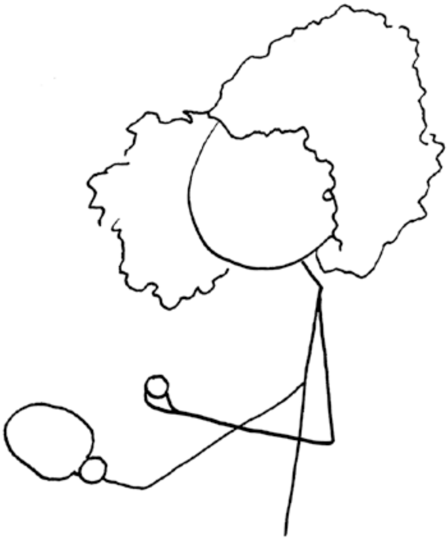

Figure 2-4. *An audio interface is like a champion ping-pong player: it does one thing and it does it extremely well*

The audio interface that I recommend is the Presonus Studio 24c as depicted in Figure 2-5.

Figure 2-5. *Presonus Studio 24c (picture courtesy of Presonus)*

I'll take you through some of its features, which will help you to use the interface effectively or to choose a different audio interface that suits your needs.

Inputs and Outputs

There are a number of different audio interfaces in the Presonus Studio Series, and model 24c has two *inputs*. An input is a pathway through which sound or audio can feed into audio equipment or computer software. The number of inputs available determines how many microphones or other sources of audio (such as a musical instrument) that you can plug into the unit at once. Depending on your needs, you can buy a model in this series with up to eight inputs. The models with four inputs or more come with expanded features and will also give you better audio quality.

The inputs are combination inputs that take microphone, instrument, and line-level signals. The sockets (Figure 2-6) are a combination of an XLR input and a quarter inch socket.

Figure 2-6. *Combined inputs of Presonus Studio 24c (picture courtesy of Presonus)*

Many audio interfaces or other pieces of audio equipment have XLR inputs and quarter inch sockets, so I will give a brief explanation of them.

XLR inputs typically look like Figure 2-7. They're important to podcasters, because they are the type of input that you plug a microphone into. As such, they are typically calibrated to receive a microphone-level signal.

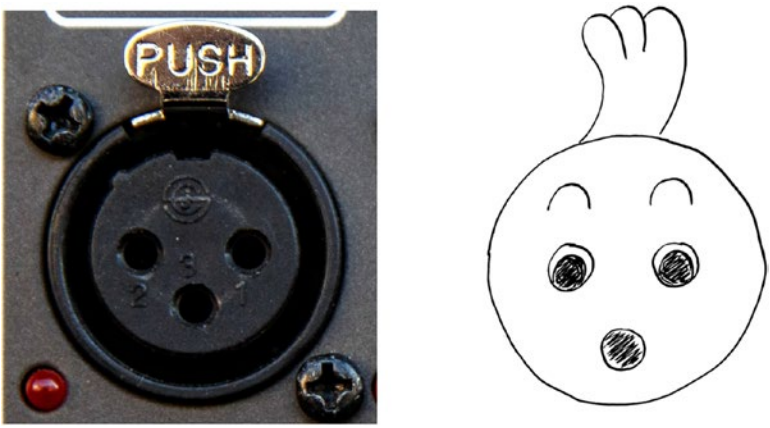

Figure 2-7. *An XLR input (Piotr Piatrouski/Shutterstock.com) on the left and on the right is what I think of when I look at an XLR input*

You'll notice that the three holes making up the eyes and mouth of the XLR input are also evident in the combined input. The hole in the middle of the combined input is a quarter inch socket, which otherwise would look something like Figure 2-8.

Figure 2-8. *A quarter inch socket (Simon Ashton/Shutterstock.com)*

The Presonus Studio 24c will receive either a *line-* or *instrument-*level signal through its quarter inch sockets. An example of a line-level signal that a podcaster might encounter is the signal coming from a mixing desk at a conference. I will discuss how you could make a recording from a mixing desk with a portable recorder in Chapter 6, "Recording Outside" (section "Plugging in to a Sound Reinforcement System"). An instrument-level signal is different again. It's the signal that comes from an instrument with passive pickups, such as an electric guitar. Many audio interfaces have a button labeled "Hi-Z" or "Instrument" next to the input which needs to be engaged when plugging in an instrument, but the Presonus Studio 24c takes care of this automatically.

The Presonus Studio 24c has phantom power which is labelled "48V." This means that you can plug in a microphone that requires external power, such as most condenser microphones. To operate phantom power, make sure that the gain on the channel is all the way down, then plug in your microphone, and then turn on the phantom power.

When you're finished using that microphone, turn the gain down, switch off the phantom power, and unplug the microphone. This practice will prevent damage to your equipment. On other interfaces, the phantom power might be labeled +48, +48V, or something similar.

Figure 2-9. *Phantom power can be "seen" by some microphones and not by others. Spooky*

The Presonus Studio 24c has two main *outputs*. An output is a pathway for audio or sound to leave a piece of equipment or software. The two outputs can function as either channels 1 and 2 or the left and right channels of a stereo signal. The outputs would most commonly be used for plugging in speakers. This audio interface also has a headphones output with a separate control.

The Audio Quality of the Presonus Studio 24c

The audio quality of the Presonus Studio 24c is great.

This interface will convert your analog audio to high-quality digital audio with a bit rate of 24 bits and the option of sample rates ranging from 44.1 kHz all the way to 192 kHz. You will know from the last chapter that this is more than enough data to make a high-quality podcast.

Another way to determine the audio quality of a piece of gear is its signal-to-noise ratio. I'll explain in terms of the audio interface. The *signal*, in the signal-to-noise ratio, is the strength of the signal coming from your microphone once it has run through the interface's preamps. The *noise* in the signal-to-noise ratio is the noise that the unit makes as it operates. It's the electronic hiss sound that you might associate with cassette tapes. As you turn up the microphone signal, you also turn up the noise of the operation of the audio interface. If there's a low amount of signal and a high amount of noise, you will have to turn up the signal so much that you can hear the noise. A quality piece of audio equipment with quality preamps will have a strong signal and not too much noise.

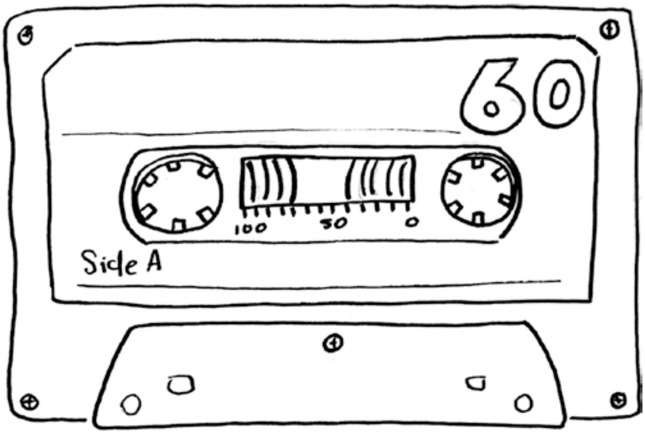

Figure 2-10. *Old cassettes have a low signal-to-noise ratio*

Audio 2-1 is a recording I made with a USB microphone with a low signal-to-noise ratio. A USB microphone is essentially a microphone and an interface built into one unit. You can hear quite a lot of noise on this track. You can also hear background noise on this track because this microphone does not reject it effectively.

Audio 2-1 Recording made with a USB microphone demonstrating a low signal-to-noise ratio

A quality audio interface will turn up your microphone to a useable level without creating a whole lot of hiss. Audio 2-2 is a recording I have made using the Presonus Studio 24c so you can judge the audio quality for yourself.

Audio 2-2 Recording made with Presonus Studio 24c

This recording sounds clean and professional. The speech is at the same loudness as the recording made with the USB microphone, but the noise is not noticeable.

Latency

The Presonus Studio 24c has low *latency*, but this is not particularly important for podcasting. Latency is the amount of delay between when audio enters and exits a system.

Latency can be introduced into a system in a number of ways. In terms of an audio interface, latency is affected by how quickly audio is converted to data and back again. Latency is affected by the specifications of your computer, including the USB ports available. The Presonus Studio 24c has the faster USB-C connectors, but also supports the slower USB-A connectors.

Latency is mostly important for podcasters when they are recording an online audio or video call. Long gaps between when a person speaks and when they are heard interrupt the flow of a conversation. When making an online call, latency will be added to your overall system due to the time that it takes to send data over the Internet. Most audio interfaces will not add a lot of latency compared to other factors such as Internet speed, but it's good to know that the Presonus Studio 24c will not be adding any noticeable amount of latency to online calls.

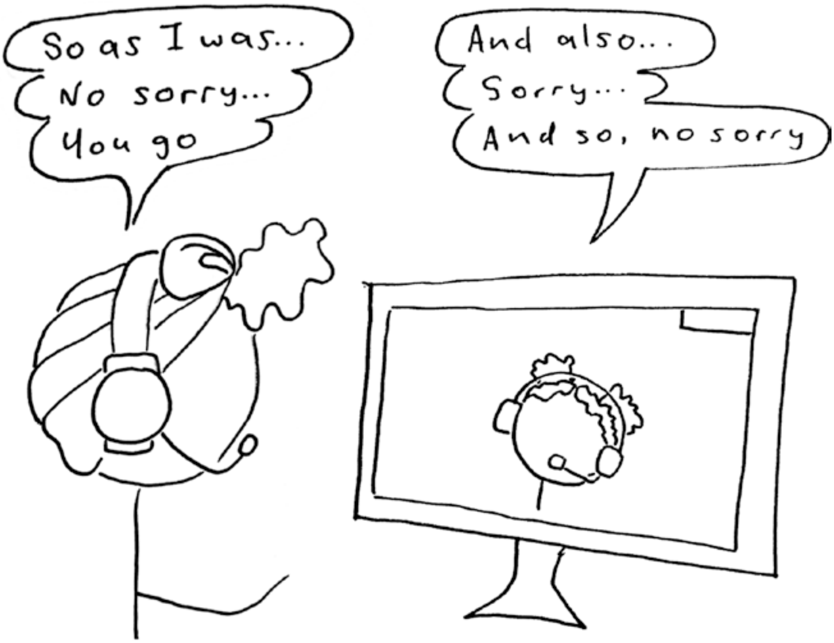

Figure 2-11. *When making online video calls, latency can cause you to lose the flow of the conversation*

You might record a podcast by gathering people together in the same room using multiple microphones plugged into an audio interface. You don't want to experience any latency in this scenario, and you don't have to. If you want to listen to your inputs with no latency, then the Presonus Studio 24c has a *direct monitoring* option. *Monitoring* is an audio nerd term for listening to audio. Direct monitoring sends your audio signal straight from the input (such as a microphone) to the output (such as headphones that are plugged into the interface) without sending it through the computer first. Most audio interfaces have a button labeled "Direct Monitor" which will toggle between the option to listen to audio from your computer and to listen directly to inputs. The Presonus Studio 24c enables you to fade between these two options with the "Mixer."

Other Considerations

The Presonus Studio 24c gets a high rating from me because it is easy to use. I plugged this audio interface into my computer, and my computer recognized it without my having to do anything else. This was not my experience with all of the interfaces that I tested; some of them required considerable effort to get started including having to install specialist software.

The Presonus Studio 24c does come with software which handles the device drivers and firmware updates. The software also allows you to change parameters such as the sample rate.

Another positive aspect of the Presonus Studio 24c is that while it performs well, it's not overly expensive for an audio interface.

There are audio interfaces available that have other features, but when it comes to choosing an interface for podcasting, the most important factor is the audio quality. From there, just make sure you choose one that has the inputs and outputs that you need.

So an audio interface takes an audio signal from a microphone, instrument, or similar; it increases the strength of the signal, converts it from analog to digital, and sends it to your computer. It also takes digital audio from your computer and converts it to analog audio so that you can play it back through your headphones or speakers. Audio interfaces are perfect for home recording setups, but not everyone makes all of their recordings at home. If you're going to be recording in the field, then you should invest in a portable recorder.

Portable Recorders

The main benefit of using a portable recorder over a dedicated audio interface is that it's portable, and so it will open up a whole world of recording outside of the home. The downside of a portable recorder is that it costs more to buy a unit that provides similar audio quality to a dedicated audio interface. This is because you're paying more for a range of extra features, and you're paying for a unit that can withstand the elements. Many modern portable recorders can also be used as an audio interface when you're recording at home. However, as portable recorders are not primarily designed to be audio interfaces, this functionality will be compromised.

The two most well-regarded portable recorder brands are Zoom and TASCAM. I've used both brands and think that they're both very good. I am particularly impressed by how hardy the units are. I know someone who had their portable recorder thrown across a park without damage. I wouldn't recommend treating your portable recorder in this way, but it's nice to know that you can. I would definitely not recommend throwing your audio interface across a park.

If you wish to make a high-quality recording with a portable recorder, then I recommend the Zoom H5 as pictured in Figure 2-12.

Figure 2-12. *Zoom H5 portable recorder*

The Zoom H5 is a great piece of kit. It's well built, fairly easy to use, and it makes a good recording. The unit has XLR inputs, meaning that you can plug one or two external microphones into it rather than relying on the inbuilt microphones. The inbuilt microphones of a portable recorder aren't that great for recording speech, but they can be used for other purposes.

Recording with the Inbuilt Microphones of a Portable Recorder

Some people record their podcast using the microphones that are built into their portable recorder. There's a certain vibe to placing a portable recorder on the kitchen table and having a yarn with your mates. However, you should be aware of some issues you will face when using this recording method.

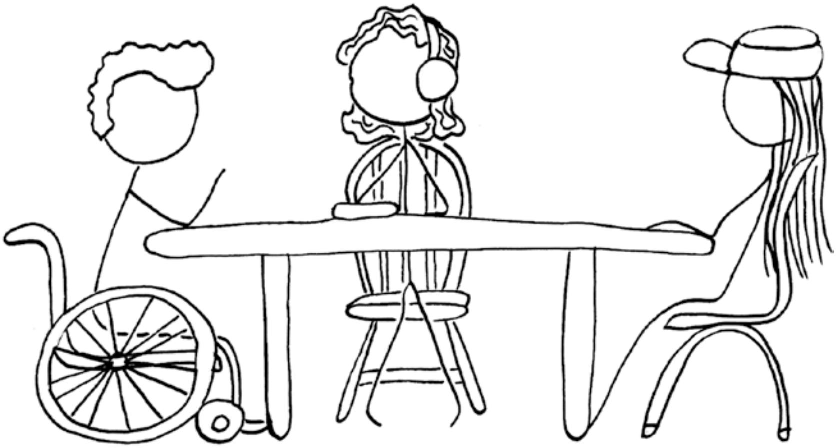

Figure 2-13. *Using a portable recorder to record yourself and your mates around the kitchen table*

First of all, the microphones that are built into a portable recorder pick up a lot of background noise. I've made a recording with the inbuilt microphones of the Zoom H5 (Audio 2-3) to demonstrate. The recording of my voice is clear enough, but you can hear my neighbors in the downstairs apartment eating lunch.

Audio 2-3 Recording with the inbuilt mics of the Zoom H5 at my home

So if you decide to record using the microphones that are built into a portable recorder, you will have to take care to avoid loud environments or accept that there will be a lot of background noise.

The bigger issue with recording using the inbuilt microphones of a portable recorder is that it is hard to get everyone at an even volume. Some of your friends will have loud voices, some soft. The people with loud voices will lean into the portable recorder, and the people with soft voices will sit halfway across the room. The presence of microphones helps people remember where they should sit, and without them, people will probably move around, making noise in the process. The recording that you're making will be on a single track, so leveling out these volume differences will be difficult and time consuming. It will be particularly difficult if there is a lot of background noise on the recording, which there will be.

There are situations where you want to record background noise. The microphones that are built into the Zoom H5 are effective for recording ambient sounds such as the soundscape of a busy city street. This is known as *atmos* to sound engineers, who love shortening words as much as they love obscure, unlistenable bands.

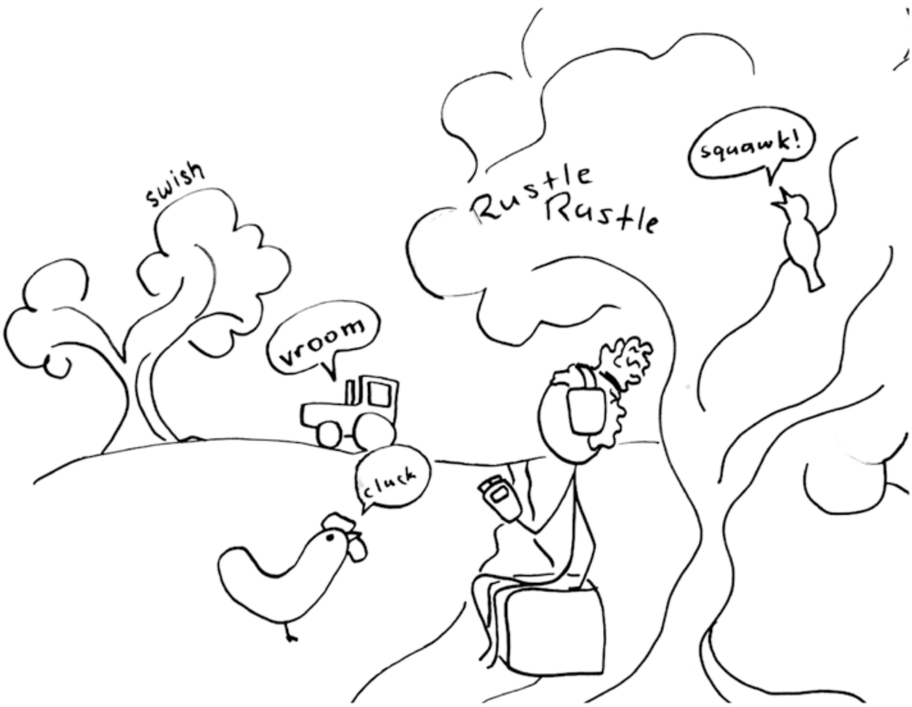

Figure 2-14. *Recording atmos at a farm*

I have made a recording of atmos using the Zoom H5 (Audio 2-4) so you can get an idea of the quality. These microphones aren't as good as external microphones, but they're decent for the inbuilt microphones of a portable recorder. The quality of these microphones is one reason I recommend this portable recorder over other units.

Audio 2-4 Atmos recorded using Zoom H5 inbuilt mics

So you can use the Zoom H5 on its own, or you can use it with an external microphone plugged into one of the inputs.

Inputs and Outputs

The Zoom H5 has two combined inputs (Figure 2-15) that take microphone-level signal and line-level signal. These inputs enable you to plug one or two external microphones into the unit, which will give you much more control over audio quality. You can also plug in other sources of audio such as the output from a mixing desk.

Figure 2-15. *The combined inputs of the Zoom H5 (photo courtesy of Zoom)*

A benefit of having two inputs is that you can record two people talking using two microphones on different tracks. This will make editing a lot easier than if you recorded them on a single track, like you would be if you were recording through the inbuilt microphones.

The inputs of the Zoom H5 have phantom power, so you can plug in microphones that require external power.

The audio quality of the microphone inputs is not quite as high as the Presonus Studio 24c audio interface, but it's pretty darn good.

The Zoom H5 records at a bit rate of 16 bits or 24 bits and a sample rate of 44.1 kHz, 48 kHz, and 96 kHz. This is more than enough for your podcasting needs.

The signal-to-noise ratio on the microphone inputs is a tiny bit lower than the audio interface, but it's still very good. Audio 2-5 is a recording I made of the Zoom H5 with a microphone at home. Listen for the small amount of noise that is evident in this track and not in the track recorded on the audio interface. You will probably need headphones to hear the noise.

Audio 2-5 Zoom H5 and external microphone

The audio quality of the inputs is one of the key reasons I recommend this model.

In terms of outputs, the Zoom H5 has a headphone output and a line output. The line output is mostly for use with video cameras. The Zoom H5 also has a built-in speaker, but I've never found that useful.

Using a Portable Recorder with an External Microphone

One major benefit of using an external microphone with a portable recorder is that you will be able to record a much higher ratio of speech to background noise.

I have made some recordings at a construction site to demonstrate the difference between recording with the inbuilt microphones of a portable recorder (Audio 2-6) and recording with an external microphone (Audio 2-7).

The speech recorded with the external microphone is much clearer, and there's much less background noise.

Audio 2-6 Recording with inbuilt microphones of Zoom H5 at a construction site

Audio 2-7 Recording with Zoom H5 with external microphone at a construction site

The Zoom H5 can record through its inbuilt microphones at the same time as it's recording through its inputs. So, if you like the background noise and want to add more later, you can record it separately using the inbuilt microphones and control how much you include.

Other Considerations

Storage and battery life are two important things to consider when buying a portable recorder.

When purchasing your portable recorder, consider buying a spare SD card with a large amount of storage. This will enable you to make high-quality recordings without having to worry about running out of space. The Zoom H5 will take an SDHC card with up to 32 GB of data. This will give you over two days' worth of mono recording for wav files that are 24 bits and 44.1 kHz. If that seems like overkill, then you can work out your recording needs with the **Audio Recording Calculator** available from the website of audio gear manufacturer *Sound Devices*.

The battery life on the Zoom H5 is pretty good. It's marketed as being able to record for five to ten hours using alkaline batteries, depending on how you are using it. It's still advisable to bring spare batteries when you're recording outside. You can run the Zoom H5 off mains power, but you'll have to buy the adaptor separately.

So a portable recorder offers portability, but it is more expensive than an audio interface of similar quality. You can use a portable recorder as is, but adding an external microphone will give you much more control over audio quality. This section is just a brief overview. I'll go into how to use the Zoom H5 in more detail in Chapter 6, "Recording Outside."

Regardless of whether you choose to use a portable recorder or an audio interface, you should always be wearing headphones when recording.

Headphones

Quality headphones are essential for both recording and editing a podcast because they will give you highly detailed and accurate information on what you are doing.

You will need headphones that cover your ears rather than ear buds. Look for a set that gives you the most honest reproduction of sound. I recommend a set without extras such as bass boost or active noise canceling.

Headphones can be either open back or closed back, and there are benefits to either design. Open-back headphones let in some noise from the outside world. This can be advantageous when conducting an interview because it allows you to better connect with the other person. It can be unfavorable in situations where you want a really accurate representation of your audio, such as editing or recording in an environment with a lot of background noise. Open-back headphones also have a more natural sound than closed-back headphones.

Whether open back or closed back, look for headphones that plug in with a cable rather than sending the signal over Bluetooth. These are more reliable, and the sound should be better quality. If your headphones do both, then make sure you use the cable when recording and editing your podcast.

Another thing to consider when buying headphones is comfort, which depends on the size and shape of your head. Durability is another important factor, especially if you will be taking your headphones on the road.

Many sound engineers use Audio-Technica ATH-M50x headphones as pictured in Figure 2-16. These have a reputation for giving an honest reproduction of sound.

Figure 2-16. *Audio-Technica ATH-M50x (photo courtesy of Audio-Technica)*

These headphones are reasonably priced, but if you already have over-ear headphones, those are probably fine for the job of recording and editing a podcast.

Cables

When you're buying your setup, don't forget to buy balanced cables with XLR connectors.

Figure 2-17. *Balanced cable with XLR connectors (photo by Karoline Morwitzer)*

Balanced cables are designed to reject electrical interference which can cause unwanted noise on your recordings. XLR connectors are standard in the audio industry. You will find XLR sockets on all your gear.

These cables can wear out or be unreliable, so it's good to own one for every microphone you are using plus one more.

Most of the time, you'll need a cable that's about three to five feet (one to one-and-a-half meters) long. This is the length of a cable that is needed to run from your audio interface or your portable recorder to your microphone.

There's no particular brand of balanced cable with XLR connectors that I would recommend, but generally speaking, if you spend more, the cable will last longer. You can get very high-quality versions of these cables, but that's not necessary. Look for the dependable "family car" version of these cables rather than the luxury sports car version.

Another cable that is useful for podcasters is the balanced cable with TRS connectors (Figure 2-18).

Figure 2-18. *A TRS connector (Anthony Maragou/Shutterstock.com)*

These cables can be used for many things including taking a feed from a mixing desk or for recording a phone call. If you're recording a phone call, you'll need one that's about 1 foot (30 centimeters) long. When recording outside, I like to bring a version of this cable that is about six to ten feet (two meters to three meters) long. This length is useful for plugging in to the back of a PA speaker and walking to the front of the speaker.

Intermission

That was the first of the two chapters on gear. It might be time to have a cup of tea so you're fresh and ready for Chapter 3, "Gear Part 2 – Microphones." I'll put the summary of both chapters at the end of the next chapter.

You'll want to be able to give this next chapter your full attention because microphones are an amazing invention that have revolutionized the world, and understanding them is essential to the process of recording a podcast.

CHAPTER 3

Gear Part 2 – Microphones

Microphones are a part of everyday life, so it's easy to overlook how they've revolutionized the world. Microphones are involved in almost every song you hear, every TV program you watch, the announcements on the train, international relations, the voice of a loved one at the end of a phone call, great political movements, the exploration of outer space, the list goes on.

One of the early uses of microphones was the telephone. It might be easy to overlook microphones today, but the potential of this invention was considered so great that it triggered a fierce patent dispute that went all the way to the Supreme Court of the United States.[1]

Microphones are also a key part of a 21st-century revolution: podcasting. As a podcaster, it's important that you understand microphones so that you can use them effectively and choose the right one for your needs. That's why this chapter contains an explanation of different aspects of microphone design.

As I mentioned in the last chapter, you will need a microphone that is designed to exclude background noise so that you can get a clear recording of speech at home or in the field. In the home environment, this will mean that you don't have to record while sitting in your closet or invest

[1] *US v. AMERICAN BELL TEL CO* [1897] U.S. 344 (United States Supreme Court).

C. M. Green, *The Podcaster's Audio Handbook*,
https://doi.org/10.1007/978-1-4842-7361-6_3

in expensive sound proofing treatments. In the field, it will enable you to record a much higher ratio of speech to background noise.

There are microphones available that will plug directly into your computer, but I have yet to come across one that compares in audio quality to using an analog microphone and audio interface. I will discuss USB microphones in more detail later.

I have tested out a number of microphones for recording a podcast, and I recommend the Sennheiser e945.

Sennheiser e945

The Sennheiser e945 (Figure 3-1) is not an expensive microphone, and as far as I know, it's never been marketed toward podcasters. It's not flashy, but it's a reliable microphone that does a good job in a variety of circumstances.

Figure 3-1. *Sennheiser e945 microphone (photo courtesy of Sennheiser)*

This is a great microphone for recording a podcast at home or in the field because its operating principle, polar pattern, and frequency response mean that it rejects quite a lot of background noise. I will explain all of these terms in the coming sections. The Sennheiser e945 is suited to recording speech because it's a vocal microphone. It is suited to recording outside because of its "rugged metal housing" (to quote Sennheiser's marketing material) and because it will operate in a wide range of climatic conditions. It has low handling noise, which is good for conducting interviews in the field.

The Sennheiser e945 rejects an impressive amount of background noise, but not all of it. Let me take you through my own home recording situation so that you can have a good idea of what to expect. I live in an apartment that overlooks the street in a noisy area, and I record in my living room which is not acoustically treated. In my own use of the e945, I find that it effectively rejects many sounds that I can hear clearly from my living room, such as people talking outside and traffic rumble from the nearby freeway. It rejects almost all of the sounds of passing cars in the street below and of the light aircraft that fly over my house. You can hear these sounds in a recording but only if you're really concentrating on it. This microphone is not so effective that it will reject the sound of a motorbike, hammering, or garbage trucks collecting rubbish directly outside my window. A category of sound which this microphone struggles to reject is squawking birds, because this is a loud sound that's fairly similar to human speech. The microphone will reject most of the bird sound, but it can sometimes be noticeable. Generally, the birds only make a racket that is loud enough to be heard on my recordings at dawn and dusk. At these times, I go into my bedroom because it has a smaller window which therefore lets in less background noise. By choosing a microphone that rejects most background noise, I only have to make small changes to my behavior to record speech at home.

In comparison to this performance, some of the other microphones I tested seemed better suited to spying than podcasting. When I listened through them, I could make out the conversations of the people in the neighboring apartments.

Figure 3-2. *A very sensitive microphone is not well suited to recording a podcast at home, but it might be useful for spying*

All of this is great, but of course, the most important thing about a microphone is how good it makes your audio sound.

This is the same audio sample as before. It is the Sennheiser e945 with the Presonus Studio 24c recording at home.

Audio 3-1 Recording made with Presonus Studio 24c and Sennheiser e945 at my home

All of the recordings made in this chapter have been made with the e945. I have repeated the audio clip of the recording made at the construction site so you can appreciate once again how well this microphone rejects background noise.

Audio 3-2 Recording made with Zoom H5 and Sennheiser e945 at a construction site

You can hear that the microphone works well, but you may be wondering why. Even if you don't intend to buy this particular microphone, the following explanation will help you understand microphones that you use in your podcasting practice.

The Benefits of Dynamic Microphones for Podcasting

The Sennheiser e945 microphone is a *dynamic* microphone. This is its *operating principle*. A dynamic microphone acts like a tiny power station, but instead of converting the wind or a waterfall into electricity, it harnesses the power of sound waves. The other main type of microphone that you will come across in podcasting is the condenser microphone, but there are many reasons why dynamics are more suited to recording at home or in the field.

Dynamic microphones pick up less background noise than condenser microphones. They also pick up less of the reverberations that are created when you make a sound in a particular space. So if you're recording in a space that acoustically is not ideal, such as a carpark, a dynamic will be a good choice. Dynamic microphones pick up less of the tiny noises of the mouth and are generally more forgiving of inexpert microphone technique. If you're taking the microphone outside, then dynamics are known to be rugged, and they can operate in a wide range of climatic conditions. For these reasons, dynamic microphones are commonly used in broadcast, both in the studio and in the field.

Some of the most well-regarded microphones in the radio broadcast industry are dynamics, such as the EV RE20, the Shure SM7B, and the EV 635A.

Condenser microphones are great, but they're not well suited to recording speech in the home because they pick up too much detail. If I was recording a talented singer in a wonderful acoustic space, I would definitely use a condenser microphone, because I would want to capture every detail of that. You could visualize this as the singer being in a bright spotlight. In terms of podcasting, I think it's better to go with the mood lighting of the dynamic microphone.

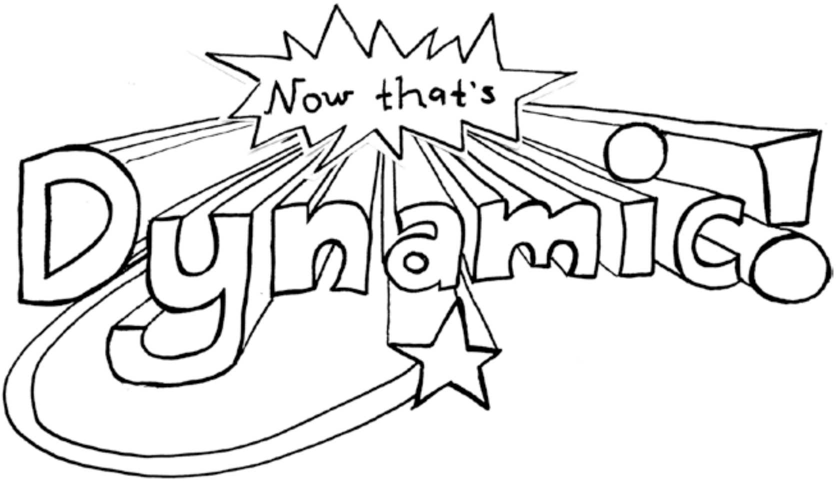

Figure 3-3. *There are many great reasons to use a dynamic microphone for recording a podcast*

So why do people use condenser microphones to record a podcast at home?

Understanding Microphone Sensitivity

The main reason that people want to use condenser microphones over dynamics is that condenser microphones have a higher *sensitivity*. The sensitivity of a microphone is a measure of the amount of signal that it makes in response to a certain amount of sound.

If we go back to our signal-to-noise ratio, a high sensitivity microphone will give you a lot of *signal*, and so you won't need to turn up your interface so much that you hear the *noise*. Another way you can create more signal is to record a louder sound, but in this case, we're recording speech. If you are using a portable recorder or an audio interface with a low signal-to-noise ratio to record speech, then a high sensitivity microphone might overcome this shortcoming.

The people who make USB microphones know this. I'll delve more deeply into USB microphones soon, but for now, I just want to talk about how USB microphones use these principles. As I mentioned earlier, a USB microphone is basically a microphone and an audio interface that has been built into one unit. If the audio interface component of this unit has a low signal-to-noise ratio, manufacturers often pair it with a condenser microphone to compensate. But, as previously discussed, condenser microphones are not well suited to recording in a noisy environment such as the home. This is one of the main reasons that I am reluctant to recommend many of the USB microphones on the market today.

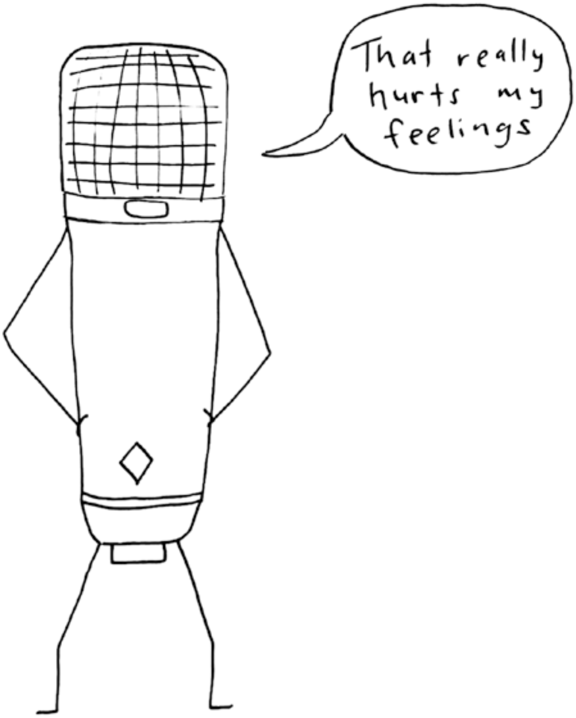

Figure 3-4. *Condenser microphones are too sensitive to record speech in noisy environments such as the home*

The Sennheiser e945 has high sensitivity for a dynamic microphone, but the sensitivity is still much lower than that of a condenser microphone. It occupies a good middle ground. It doesn't pick up too much background noise, but it is sensitive enough to record speech using the audio interface or the portable recorder mentioned previously with very little noise from the operation of the unit.

If you are using equipment with a low signal-to-noise ratio, a condenser microphone will compensate, but a condenser microphone picks up much more background noise, room reverberation, and mouth noise. Therefore, using a high-sensitivity dynamic microphone such as the Sennheiser e945 is a better option for recording a podcast at home.

Choosing a Microphone with an Appropriate Frequency Response

When you are choosing a microphone, it is important to match the frequency response to your intended use. The *frequency response* refers to how well a microphone picks up certain frequencies and rejects others.

Frequencies are a bit like the pitch of a sound. An example of a low-frequency sound is the sound of a bass guitar, and an example of a high-frequency sound is the sound of a whistle. There are microphones that are designed to favor certain frequencies, such as microphones that favor low frequencies and so are used to record kick drums.

The frequency response of the Sennheiser e945 microphone is tailored for vocals. This is why it's referred to as a *vocal microphone* in its marketing materials.

A vocal microphone favors the frequencies of the human voice. This means that it rejects more of the frequencies outside of this range, thus reducing the amount of background noise on your recording. For example, I've found that the Sennheiser e945 microphone is good for rejecting the traffic rumble that I can hear in my living room. The particular frequencies that this microphone enhances also enable it to cut through background music or sound effects better than some of the other microphones I tested.

Polar Patterns and How They Help Reject Background Noise

Different microphones have different *polar patterns*. The polar pattern of a microphone determines the areas from which it picks up sound most effectively and the areas from which it rejects sound. Understanding the polar pattern of your microphone will help you position it for maximum effect. Omnidirectional, cardioid, supercardioid, and figure 8 are polar patterns that you might encounter when recording a podcast.

Polar patterns are represented by the following symbols throughout the audio industry. Look for them engraved into the body of your microphone.

An *omnidirectional* microphone will pick up sound from any direction. This is represented by a circle (Figure 3-5), but really the microphone is picking up sound in a spherical area.

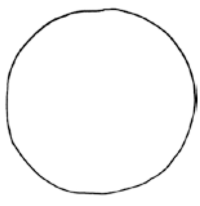

Figure 3-5. *The symbol representing an omnidirectional polar pattern*

Any microphone that's not omnidirectional can be described as *directional*. Directional microphones are designed to pick up sound from certain directions and reject it from others. If you're recording with a directional microphone, you should point it away from any source of unwanted noise.

The benefit of using a directional microphone is that you will pick up less background noise. The downside of having areas where the microphone rejects sound is that sound that does come through these areas can be not quite right. This is referred to as *off-axis coloration*. Many people use omnidirectional microphones in the field to avoid off-axis coloration.

The most common type of directional microphone is a cardioid microphone. This picks up sound from the front and sides. Its symbol is this bum shape, as shown in Figure 3-6.

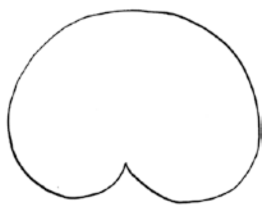

Figure 3-6. *The symbol representing a cardioid polar pattern*

Microphones with a cardioid polar pattern are good at rejecting background noise, but not as good as supercardioid microphones.

The Sennheiser e945 is supercardioid. This means that it picks up the most sound from the front and the sides and also a little bit from the back.

The symbol for a *supercardioid* mic is as in Figure 3-7, but it's better explained with Figure 3-8.

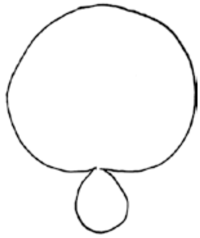

Figure 3-7. *The symbol representing a supercardioid polar pattern*

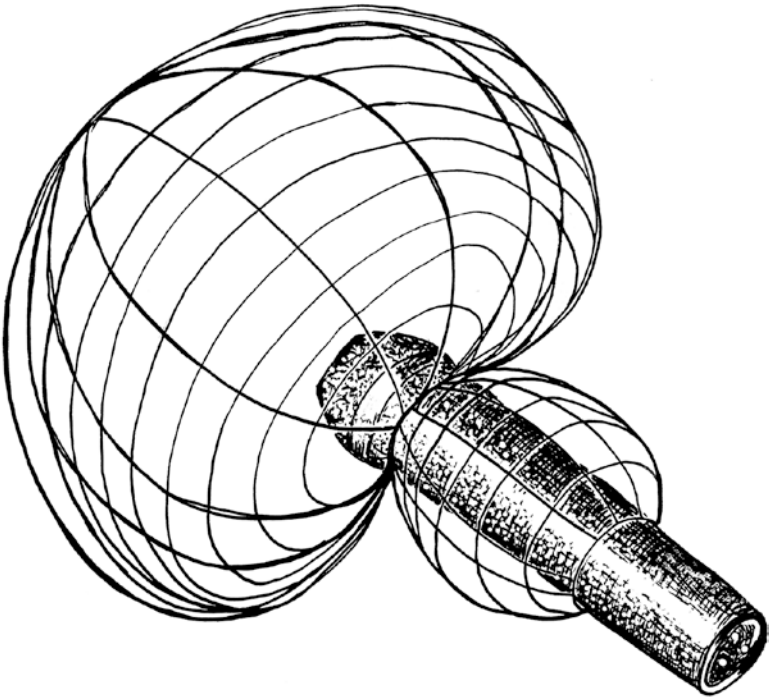

Figure 3-8. *The approximate area around a supercardioid microphone where it bests picks up sound*

A supercardioid microphone rejects quite a lot of background noise. Most of the time, it is not a big deal that there is an area at the back that picks up sound, but it can be a problem if there's a source of unwanted noise on the ground, which is an unusual scenario for podcasters.

A microphone that picks up sound from the front and back has a *figure 8* polar pattern, as in Figure 3-9. These are really good at rejecting sound from the sides.

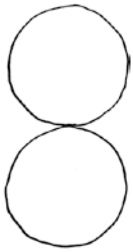

Figure 3-9. *The symbol for the figure 8 polar pattern*

Whichever microphone you use, look for the polar pattern symbol engraved in the handle. Better yet, have a look at the manual of your microphone to get a more detailed picture of the directions from which it will pick up and reject sound. An even more effective strategy is to move the microphone around a source of noise and listen to how it responds.

I explore the topic of polar patterns in more detail in Chapter 6, "Recording Outside," in the section "Background Noise."

Now that you understand some of the key features of microphones, let's discuss USB microphones.

USB Microphones

USB microphones are a technology that combines an audio interface with a microphone in one package. Many people use them for podcasting.

An advantage of using a USB microphone is that they are easy to understand. They are a bit like your computer mouse – you pull them out of the box and plug them straight into your computer, and usually you can get a signal out of them straightaway. They're a lot cheaper than purchasing a separate microphone and interface or a microphone and portable recorder.

A drawback of a USB microphone is that you can use only one microphone at a time because currently computers recognize only one interface.

USB microphones are not usually compatible with portable recorders and so are not suited to recording in the field. An exception to this would be a model with an XLR output as well as the USB output.

Many USB microphones are condensers, so if you are looking for a USB microphone, consider the many drawbacks of recording with a condenser in the home or in the field. If you are trying out a dynamic USB microphone, then listen for noise.

I don't want to be a snob or a traditionalist, but it is my opinion that this technology has yet to meet the needs of podcasters. I do think that in the next few years there will be a USB microphone that is well suited to podcasting.

Microphone Stands

If you are recording at home, a microphone stand will improve the quality of your recording. It will prevent the sound of handling noise for any microphone. It will help you keep the microphone in a good position, and in a consistent position, without your arms getting tired.

There are two types of stands to consider: the standard desktop stand and the scissor arm boom stand.

There are a few different standard desktop stands (Figure 3-10) available. These all tend to work pretty well. You'll have to make your own judgment on the quality of the parts in comparison to the cost.

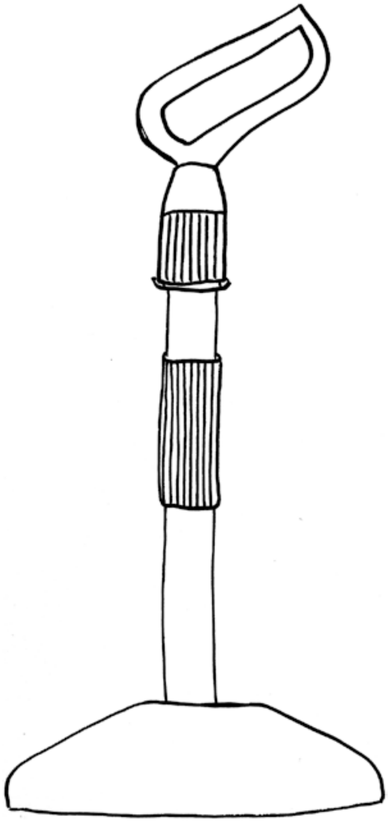

Figure 3-10. *A table microphone stand*

A desk-mounted scissor arm boom stand (Figure 3-11) is good for getting the microphone in just the right position. However, it has many more moving parts than a desktop stand, and so the quality and the cost vary considerably. I personally use a bottom-of-the-line model and it works OK, but it looks like it will have a short life. When buying one of these stands, try it out with the heaviest microphone you intend to use and see how it performs. Spending more money on a quality item should buy you a model that lasts for longer.

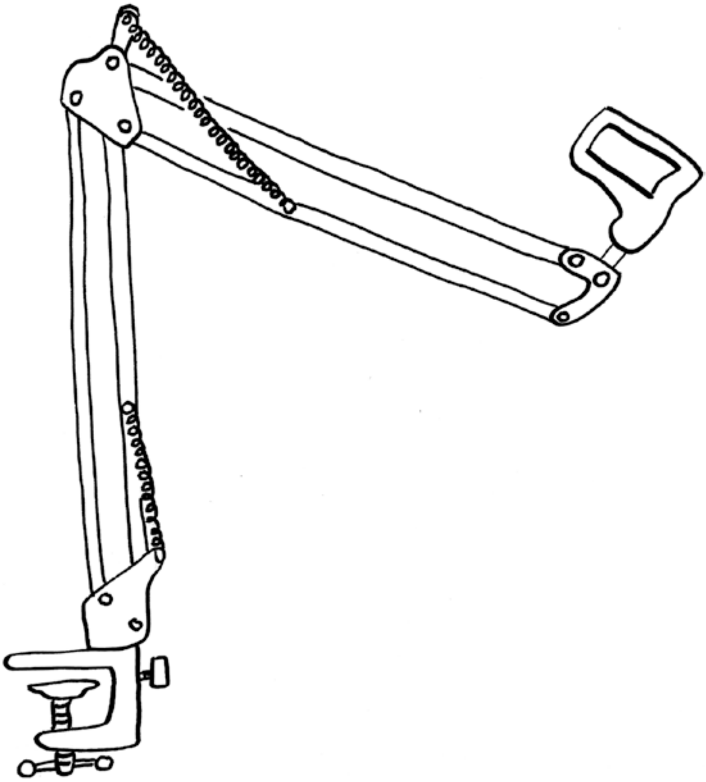

Figure 3-11. *A desk-mounted scissor arm boom stand*

Spider Shock Mount

If your microphone stand is on or attached to the same desk as your computer, then your microphone might pick up vibrations. Use a spider shock mount to isolate the microphone from your computer.

The spider shock mount goes on the end of a microphone stand in the place of the microphone clip. There are a few different designs, but the basic idea is that the microphone sits in the middle and the rubber bands work to isolate it from shocks and vibrations.

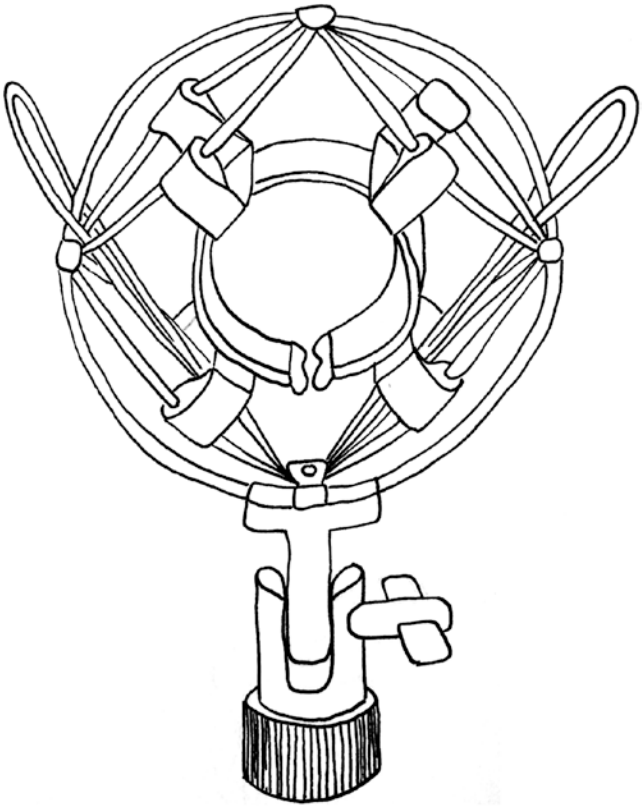

Figure 3-12. *A spider shock mount*

You'll need one that's the right size for your microphone. If shopping in store, test it out with your microphone; if shopping online, include the model number of your microphone in the search term.

Mic Sock

The mic sock is the little hat that goes on the microphone. For reasons that are beyond my comprehension, they don't come as standard with all microphones, even though they are cheap and useful. A mic sock keeps your microphone clean which will extend its life. There's no particular brand of microphone sock that I recommend as they are all pretty much the same.

Figure 3-13. *A mic sock (photo by Karoline Morwitzer)*

There are special mic socks that are used for recording outside. These are called *dead cat* (apologies, I know it's not a great name) or *windjammer* wind socks. These are not as cheap as regular mic socks, but good ones improve the quality of outside recordings a great deal because they stop wind noise. The quality of these vary, and I recommend the RØDE Deadkitten (Figure 3-14). Apologies for that name too. The RØDE Deadkitten fits over the Sennheiser e945 and over the inbuilt microphones of the Zoom H5, along with many other microphones.

Figure 3-14. *RØDE Deadkitten wind sock (photo by Karoline Morwitzer)*

Summary

Good audio quality requires an investment in the right kind of gear for your recording environment. In return for your investment, you will find that the recording process is easier and the result is better. You can make substantial improvements to your sound without investing in gear, but you're never really going to achieve a high-quality recording at home without investing in quality gear that is suited to podcasting. If your budget is tight, consider hiring a studio or hiring equipment.

Recommended equipment for recording inside:

- Audio interface – Presonus Studio 24c
- Microphone – Sennheiser e945
- Headphones – Audio-Technica ATH M50x
- Balanced cable with XLR connectors

- Balanced cable with TRS connectors (if recording phone call)
- Microphone stand
- Spider shock mount
- Microphone sock

Recommended equipment for recording outside:

- Portable recorder – Zoom H5
- Microphone – Sennheiser e945
- Headphones – Audio-Technica ATH M50x
- Balanced cable with XLR connectors
- Balanced cable with TRS connectors (if taking a feed from a mixing desk)
- Wind sock – RØDE Deadkitten

CHAPTER 4

Getting a Good Take

When I started making recordings, it felt like the time I spent editing could be measured on a geological timescale. Children grew, then went on to have children of their own. A single seed flourished into a mighty forest. Civilizations fell and new ones arose in their place. The acceptable position of the waistband of trousers went from high to middle to low and back to high again. Still I was at my computer, editing.

The reason that I spent so long editing is that I was trying to fix bad recordings. This experience has led me to deeply appreciate that audio which has been recorded correctly sounds better and is much easier to edit.

This is not just my opinion, it is a well-respected principle in the field of sound engineering. I would like to share this popular sound engineering meme with you (Figure 4-1).[1]

[1] memegenerator.net. 2021. *batman slap robin - Let's just fix in the Mix... ....* [online] Available at: https://memegenerator.net/instance/82085483/batman-slap-robin-lets-just-fix-in-the-mix-record-it-properly [Accessed 7 September 2020].

C. M. Green, *The Podcaster's Audio Handbook*,
https://doi.org/10.1007/978-1-4842-7361-6_4

Figure 4-1. *A popular sound engineering meme (memegenerator.net)*

Much of this chapter focuses on the act of putting a microphone in front of someone's face. This sounds simple enough, but there are many perils to avoid: not recording enough audio data, peaking, sibilance, popping, mouth noise, shuffling, rustling, the list goes on. These recording errors can be a distraction from what a person is saying, and they might even render valuable audio unusable.

I strongly encourage podcasters to take the time and effort to get the recording stage right, and that will be the focus of this chapter. Saying that, when conducting an interview, there are two things more important than the quality of the audio: the quality of the content and the needs of the people involved. You want to look after the people you're recording because it's the right thing to do and also because the needs of the people involved feed straight back into the quality of the content. Look after the people you're recording, and you will be rewarded with a better end product. I have every faith that you can balance all of these considerations: audio quality, content, and the needs of the people involved. While a bad recording of good content is still better than a good recording of filler, a good recording of good content is a winner!

Sound Check... Check... 1... 2...

Audio advice is a framework to help you get started. What's really important is how a recording sounds. As such, you should perform a sound check before making a recording.

Whether you're recording at home, in the outside world, or over the Internet, you should ideally set up your equipment and test it in that space first. Then you can start the sound check.

When you tell people that you're performing a sound check, they tend to sit up straight and talk right into the microphone in either an unusually loud or soft voice. They might even get the urge to start saying "check... 1... 2..." because they've seen sound engineers tuning a PA before a live event. None of this is going to give you any useful information for setting up your equipment. This is because once you start the recording people will gradually move to a more comfortable position away from the microphone and return to speaking at their usual volume.

To perform a sound check, you need to get a person to sit comfortably and then bring the mic to them. This is where it's useful to have a desk-mounted boom stand, so that you can get it in just the right position.

A sound check can make people feel awkward, which then makes them speak in an unnatural way. The trick to making a person speak at their normal volume is to ask them a question which makes them think a little bit. Personally, I like to lead with "What did you have for breakfast?" and go from there. As a consequence of this practice, I've discovered that you can neatly divide Australian politicians into two camps: a very healthy breakfast or no breakfast at all. This trick isn't just for politicians; listen to Audio 4-1 to hear the breakfast choices of veteran peace activist Graeme Dunstan.

Audio 4-1 Performing a sound check

Now you've had a chance to observe the person whom you're recording while they're speaking, and you might want to move the microphone. Here are some things to look for: does the person have notes that they're going to refer to? In this case, move the microphone halfway between the head position they'll have when they're looking at the notes and the head position they'll have when they're looking at you. Similarly, if you are recording multiple people, you might want to adjust the position of the microphones to allow them to slightly turn their heads to look at each other and back at you.

Keep the conversation flowing as you make these adjustments so that you can hear the effect of any new microphone positions. If you can make people laugh, that is excellent. Laughter tends to be about 10dB louder than normal speech, so it is an important check.

Figure 4-2. *Asking someone what they had for breakfast will take their mind off a sound check which will help them speak in a more natural way*

Listen through your headphones for little sounds. Small, persistent sounds can distract a listener from your content, and it's time consuming to edit them out. If the person you're recording is wearing a watch or a bangle that's hitting the desk, ask them to take it off. Similarly, remove small objects that they might fiddle with, such as a pen. Sometimes the person you're recording might tap on the desk or thump it when they've made a particularly good point. Feel free to tell them to stop. Now is also the time to ask people to turn off their phones or put them in airplane mode. It's better to catch these things in sound check, but don't worry too much if you've missed something. If the recording has started and it seems appropriate, you can still remedy these problems because podcasts are prerecorded, so you can just edit that bit out. Even if you wait until the end of a person's thought to ask them to remedy an audio problem, you will only be editing a few minutes of undesirable audio, rather than a whole interview's worth.

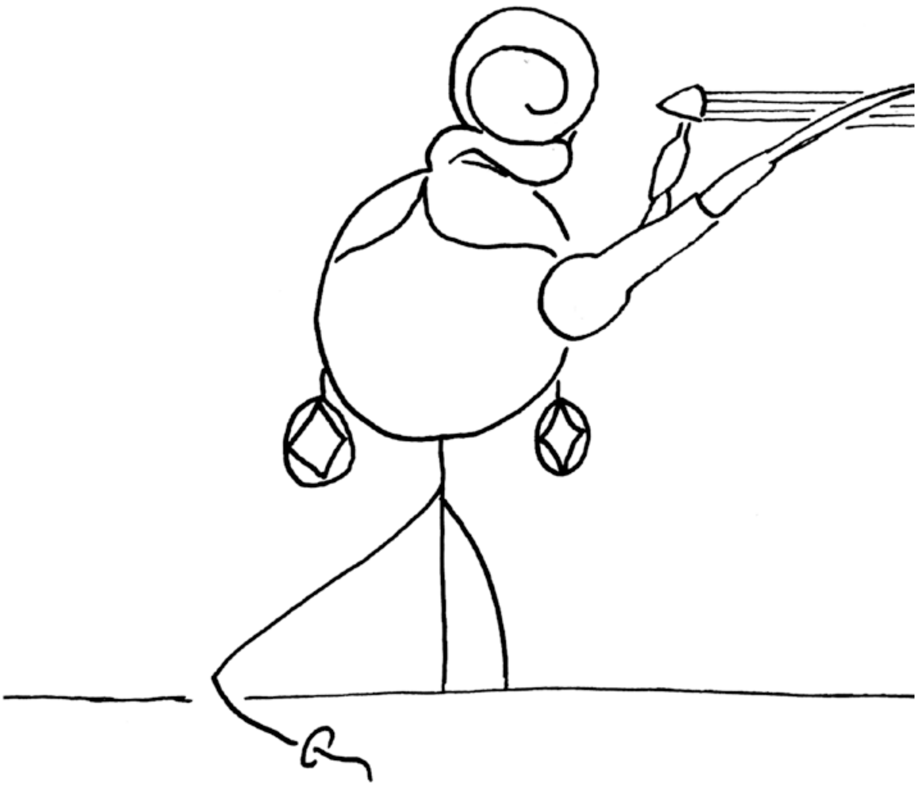

Figure 4-3. *Asking a guest to remove a piece of jewelry that is hitting the desk might compromise their outfit, but it will save a lot of time in the editing process*

Having said this, you might consider these little sounds a part of the storytelling. People fiddle with things when they're nervous. The sound of tapping on a desk could be distracting, or it might help communicate to your listener the emotional state of the person who's talking. I edited a recording recently where a person was shuffling paper, and that sound communicated to me that they'd done a lot of research. Use your judgment as to what you think is appropriate.

You should always be wearing headphones when recording, and if possible, you should set up everyone else with a pair as well. Headphones will help people to catch themselves making noises that are not ideal for the recording.

When I started out as a journalist, I was a bit embarrassed to ask people to move or to let me adjust the equipment too much so that I could get a good recording. Eventually, I realized that if a person had agreed to let me interview them, then they probably wanted a good recording as well. A good recording will help them in their presentation of whatever point of view they're putting forward. Obviously, there are exceptions to this, and you'll have to use your good sense and communicate with the interviewee. A person who's very upset about something might have less tolerance for fussing than a person who's selling a book. However, a person who's very upset might be deeply invested in getting the audio quality right. The fussing around with equipment could make them feel reassured that you know what you're doing and that you're taking their story seriously. You're going to have to read the room on this one.

No matter what the situation, you'll need to do some kind of sound check. It's wildly embarrassing to have to go back with cap in hand and admit that you made a completely unusable recording and that you have to do it again.

Positioning the Microphone for Recording Speech

An important part of a sound check is getting the position of the microphone right. There are two key considerations for setting the position of a microphone: you want it to sound good and you want everyone to be comfortable. In this section, we will consider the distance the microphone is from the mouth, the angle that the microphone is set at, and some common issues that arise from the incorrect placement of a microphone. This section assumes that you are using an external microphone, preferably a dynamic vocal microphone, rather than the inbuilt microphones of a portable recorder. For more information, see Chapter 3, "Gear Part 2 – Microphones."

So, if you are recording speech using an external microphone, aim to position the microphone about the distance of a closed fist away from the speaker's mouth on about a 45° angle as in Figure 4-4. Use the fist of whoever you are recording to take the measurement. You want the microphone to be pointing at the mouth, but out of the way of the breath. Further adjustments can be made from here. This microphone position is common practice for recording speech for radio or podcasting, but it is not useful for other applications such as recording a singer in a studio.

Figure 4-4. *Start by positioning the microphone about a fist's width away from the mouth on approximately a 45° angle*

Once you have positioned the microphone, turn the gain knob up until you have a healthy volume level. Gain is a bit like volume. I will go into further detail about gain later in this chapter, but first we will consider the angle of the microphone.

Setting the Best Angle for the Microphone

You started the sound check by positioning the microphone at about a 45° angle pointing at the mouth. The angle of the microphone will need further adjustment if you can hear *sibilance* or *plosives*. These are common issues that arise when recording speech. In case you're not familiar with these terms, I've created some examples.

Sibilance is the exaggeration of "s" and "f" sounds as in Audio 4-2.

Audio 4-2 Sibilance

Plosives are similar to sibilance, but are usually related to the letter "p." Plosives are sometimes called *popping* because they are accompanied by a distinctive "popping" sound as in Audio 4-3.

Audio 4-3 Plosives

Understanding the cause of sibilance and plosives will help you prevent them. A microphone has a thin membrane inside of it called a diaphragm that vibrates in response to sound waves. Vibrations of the diaphragm are converted into fluctuations in electrical pressure (voltage). This is the process of turning sound into audio. If you are using an *end-address microphone*, that is, a microphone where you talk into one end instead of the side, then the diaphragm is positioned as in Figure 4-5.

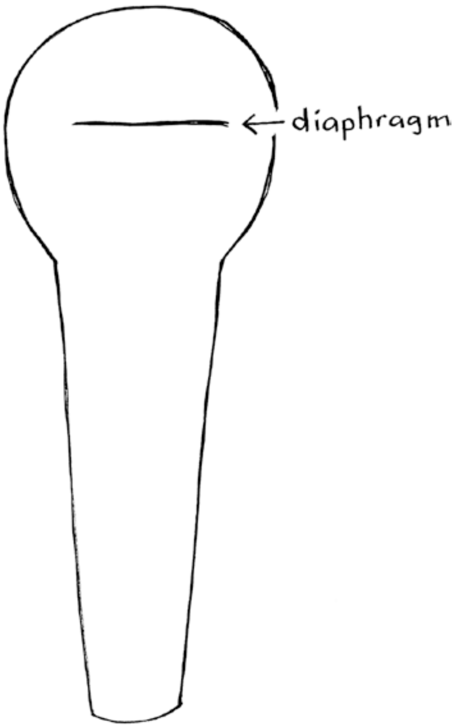

Figure 4-5. *The position of the diaphragm in an end-address microphone*

When you're speaking directly into a microphone, the sudden rush of air caused by certain consonants can cause the diaphragm to move a lot, thus causing those consonants to be accentuated.

To prevent the diaphragm from moving too much, position it at a 45° angle, out of the way of the breath, as in Figure 4-6. You can think of it like this. When you're flying a kite, you want to position it to catch the wind. When you're using a microphone, you want to do the opposite. A direct blast of air straight on to the diaphragm will move it the most, but if you angle the microphone, then you reduce this effect. If angling the microphone isn't helping, try moving the microphone to the point where you can hear the most sibilance or plosives and then move it away from there. The microphone does not need to be below the mouth, it can be at a 45° angle coming from any direction.

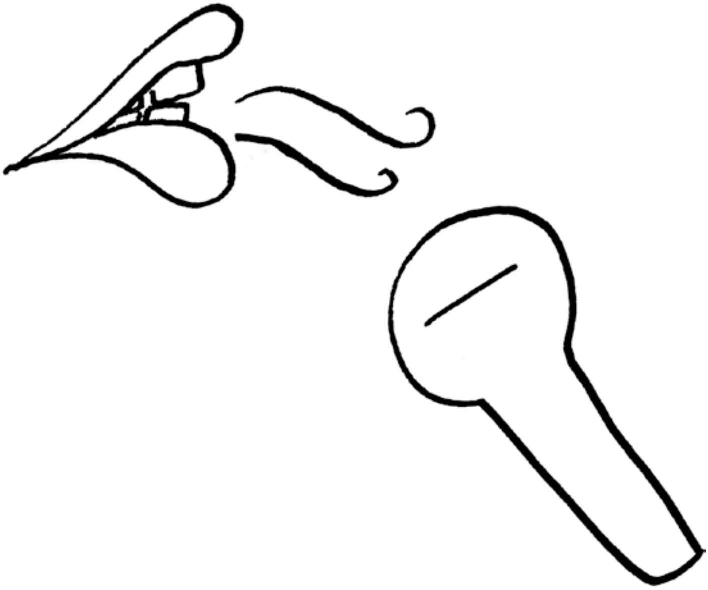

Figure 4-6. *To reduce sibilance or plosives, angle the microphone about 45° away from the speaker's mouth. It should be pointed at the mouth but below the breath*

There are a few other ways to reduce sibilance and plosives. Your microphone choice will influence the amount of sibilance that you are recording. Condenser microphones are more sensitive to sibilance than dynamic microphones.

A mic sock will help with sibilance and plosives a little bit.

There's a piece of equipment called a *pop filter* that is designed to reduce plosives. If you're having a particular problem with plosives, you can purchase one, but I find that concentrating on your microphone position is a cheaper and easier solution. Pop filters tend to be used when people are singing, which is not the same as recording speech.

Positioning the microphone at an angle will help you to prevent the recording of plosives and sibilance, but you should be mindful to stay within the area in which the microphone picks up sound most effectively.

Staying On-Mic

When recording a person speaking, you should take care that they stay *on-mic* as opposed to *off-mic*. If a person sounds *off-mic*, it means they're talking into an area of the microphone that does not pick up sound effectively. I gave a brief explanation of polar patterns in Chapter 3, "Gear Part 2 – Microphones" in the section entitled "Polar Patterns and How They Help Reject Background Noise". If you haven't read that section, I recommend that you read it now.

Audio 4-4 is a recording which demonstrates how on-mic sounds compared to off-mic. A person who is off-mic sounds distant and muffled. A person who is on-mic is coming through loud and clear.

Audio 4-4 On-mic and off-mic

Headphones help most people stay on-mic, but some people become very animated when they speak. If you are recording someone who won't stay on-mic, I've found it helpful to let them have an experiment with what's on-mic and off-mic before starting the recording.

Having the microphone at the right angle will help you avoid sibilance and popping. Having it at the right distance from the speaker's mouth will help these issues and reduce the recording of mouth noises.

Setting the Distance Between the Microphone and the Speaker

At the start of the sound check, you positioned the microphone about the distance of a closed fist away from the speaker's mouth. The fist is a good guide because bigger people tend to have both bigger fists and bigger voices. After you have listened to the person speaking, you might need to make further adjustments to the distance between microphone and mouth to improve the recording.

Changing the distance of the microphone from the speaker can resolve some technical issues. If a microphone is too close, you might run into some problems: sibilance, plosives, or mouth noise. Move the microphone a bit further away, and turn the gain up. If there's too much background noise, move the microphone closer, and turn the gain down. Hopefully, the background noise will cover any mouth noise.

The distance of the microphone from the speaker will change the character of the recording. If you move the microphone further away, you're not just recording more background noise, you're recording more of the *reverb*. That is, you're recording less of the direct sound and more of the repeats of the direct sound that have returned after bouncing off other objects such as the desk and the walls of the room. From a storytelling perspective, a recording that's made with a microphone that's further away from the speaker sounds like a person in a room, whereas a recording that's made with a microphone that's very close to the speaker sounds authoritative. The latter microphone position might be good for a *voice-over*, which is a recording of a person speaking that is intended to be explanatory or to tie different sections of a podcast together.

It's up to you to judge which is the best balance between all of these factors for your particular recording. I mentioned mouth noise as a sign that you have the microphone too close to the mouth, but mouth noise can be influenced by other factors.

Avoiding the Recording of Mouth Noises

When you're having a conversation with someone, you're usually doing it from a safe distance, so you don't notice all of the little noises of the mouth. If a microphone is too close to the mouth, then these noises will be magnified, which sounds… organic. Microphone placement is just one way to prevent the recording of mouth noises.

Sometimes people have very dry mouths, and their mouths make an unusual amount of noise. In my work as a journalist, I have learned that not only are our politicians skipping breakfast, but some of them are not even adequately hydrated. The number one solution to a dry mouth is to kindly suggest to the person you're recording that they need to drink a glass of water. After that, you need to engage them in conversation for a short time so that their mouth doesn't sound super wet on the recording. This whole situation may be awkward, but it's less awkward than a podcast full of mouth noises.

Your choice of microphone will affect the recording of mouth noises. Condenser microphones are more sensitive than dynamics, and so will pick up more tiny noises.

If you've done your best and there's still mouth noises on the recording, then you can fix them by going through your recording and deleting them. Some will be easier to delete than others. This process is laborious and best avoided.

Positioning the Microphone for Different Types of Recordings

Now that you know what to listen for, you should be able to record speech clearly and cleanly. This needn't be an overly complicated process. It comes down to this: for recording speech, place the microphone about the distance of a closed fist away from the mouth of the speaker, at about a 45° angle, and then adjust it until it sounds right. The microphone should be pointing at the mouth and out of the way of the breath. If you've got that right, then you're on a winner, but you can take it further. As I previously stated, you can position the microphone so it points at the mouth from a 45° angle from almost any direction: below the mouth, above the mouth, off to the side. I am going to discuss a few options and how they will produce different results.

When recording a voice-over, it is good to position the microphone above the level of the speaker's mouth pointing down as in Figure 4-7. This will encourage the speaker to tilt their head up and open their airways. This is quite a relaxed speaking position which should result in a nice, clear voice. When you are positioning the microphone above the level of the speaker's mouth, you don't need to place it right in front of their eyes; you can position it off to the side so they can still see.

Figure 4-7. *When recording a voice-over, position the microphone above the level of the speaker's mouth pointing down*

When you have the microphone in this position, you'll need to position any notes or script in front of the person to encourage them to look forward. Having the notes on a computer screen is a good way to do this. It might even be worth propping the computer up on a few books. This microphone position is good for many types of recordings, but it would block eye contact between an interviewer and interviewee.

In situations where eye contact is critical, such as conducting an interview, then you should position the microphone underneath the speaker's mouth, straight on, pointing up. See Figure 4-8.

If you want your recording to have a lot of energy, consider positioning the microphone so that the person who is speaking is standing up. This might be good for your podcast of a Shakespearean play. The Australian radio service SBS goes so far as to have motorized desks in its recording studios to accommodate this, but you don't need to go to that much effort. You'll just need a microphone on a desk-mounted boom stand and printed notes.

Recording speech is as simple as setting up your equipment, positioning the microphone, and pressing record. Recording speech to a high quality is almost that simple; it just requires a little more care.

Figure 4-8. *In situations where eye contact is critical, position the microphone underneath the speaker's mouth, straight on, pointing up*

Setting Levels

Earlier I referred to setting the level of a recording during the sound check. There are a few terms for setting your levels. There's *gain*, *volume*, *level*, and *balance*. These terms are basically interchangeable, except for gain. Gain specifically refers to *amplifying* a signal, that is, making a signal stronger. You can use *gain* to set your *level*, but not *level* to set your *gain*.

You will find gain controls on your audio interface (Figure 4-9), your portable recorder (Figure 4-10), and in your audio editing software (Figure 4-11).

Figure 4-9. *The two gain knobs on the Presonus Studio 24c audio interface*

In Figure 4-9, I have circled the two knobs that control gain on the Presonus Studio 24c audio interface. The one marked "1" is for channel one, and the one marked "2" is for channel two.

If you think two gain knobs is good, the Zoom H5 has three. One for channel one, one for channel two, and one at the top of the unit that controls the inbuilt microphones. See Figure 4-10.

Figure 4-10. *The gain knobs on the Zoom H5 portable recorder*

Audacity has a gain control in the panel to the left of every track of audio as depicted in Figure 4-11.

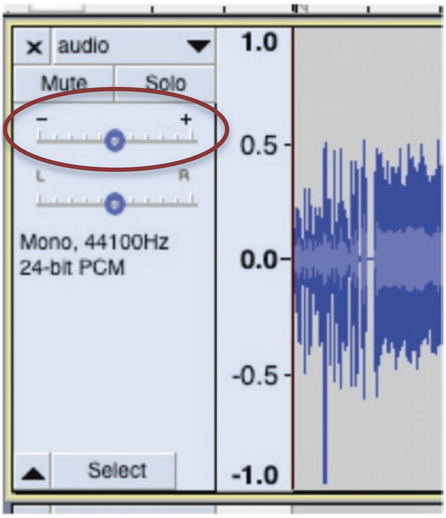

Figure 4-11. *The gain control on a track in Audacity software*

If you are recording into a computer using an audio interface, then you should preference the gain control on the interface over the gain control in your audio editing software. One of the main reasons to invest in a decent audio interface is its superior ability to amplify a signal. Saying that, you might not be using a decent audio interface, so if it sounds better to preference the gain control on the computer, then by all means use that gain.

So the gain control can be used to set your level. In Chapter 1, "File Formats and Settings," I stated that setting the right bit rate will enable you to make a high-quality and detailed recording. Setting the correct bit rate is half of the story, and the other half is recording at the right level.

The crux of this section is that when it comes to setting your level, a higher level will mean that you are recording more audio data. If your level is too low, you will not be recording enough audio data. If you set your level too high, the audio will peak and it will be distorted. So you need to find that nice middle ground at which to set your level.

Recording Sufficient Audio Data

To make a good digital recording, you will need to set your level high enough to record an adequate amount of audio data. I will discuss the exact specifications of this later on.

There are a number of reasons why you need to record enough audio data. Audio data is what provides clarity and detail to your recordings. Audio data is lost through editing and compression processes, so you want to start with a good amount. Certain effects that you might add such as reverb rely on having a decent amount of audio data to work effectively.

If you record something at a low level, then you can turn it up in the editing process, but you won't be adding any audio data. This is similar to when you're working with a digital image on your computer. If you take a small image and make it larger, the computer will try to fill in the blanks, but it will never have the same amount of detail as a large picture that was taken at a high resolution.

Figure 4-12 is a photograph I took of ibises, a bird found in Australia. I resized this picture to be small, saved it, and then tried to restore it to its original size. You can see that the computer made a valiant effort to fill in the blanks, but has not quite succeeded. This is the same thing that happens when you record something quietly and then turn it up in your editing program.

Figure 4-12. *Resizing a small digital image into a large digital image*

You can, however, take a large digital image and reduce the size, and it will look fine. The computer will simply delete unnecessary data. The same is true for recording – it's better to record at a high level and turn it down later than to record at a low level and turn it up.

Another risk of recording at a low level is that you will hear the noise floor of your recording equipment. The noise floor sounds like the electronic hiss that you might associate with cassette tapes. The recording might sound fine as you are making it, but when you later increase the volume, you will also increase the volume of the noise floor.

A common mistake people make when recording is to turn the headphones volume up when they mean to turn the gain up. Turning the headphones up might make the audio sound good to you, but it will not necessarily result in a good recording.

It's good to record a decent amount of audio data, but you don't want to set the level so high that you're peaking.

Peaking

Peaking, also known as *clipping*, occurs when you set your level too high.

If your audio file is turned up so loud that it is touching the top of the track, then it is peaking. See Figure 4-13.

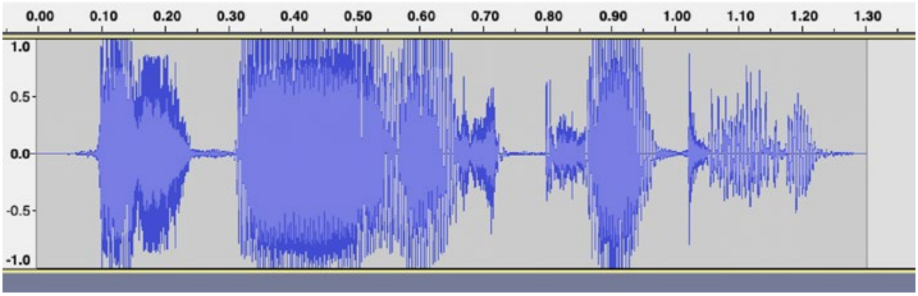

Figure 4-13. *An audio file that is peaking (Audacity software)*

The end result of peaking is distortion. You can hear how that sounds in Audio 4-5.

Audio 4-5 Peaking

A distorted recording of speech sounds bad, so it is important to leave enough *head room* to avoid it. This is the amount of space between the top of the loudest part of the waveform and the top of the track in the audio editing program. Figure 4-14 is my artistic interpretation of the concepts of *head room* and *peaking*.

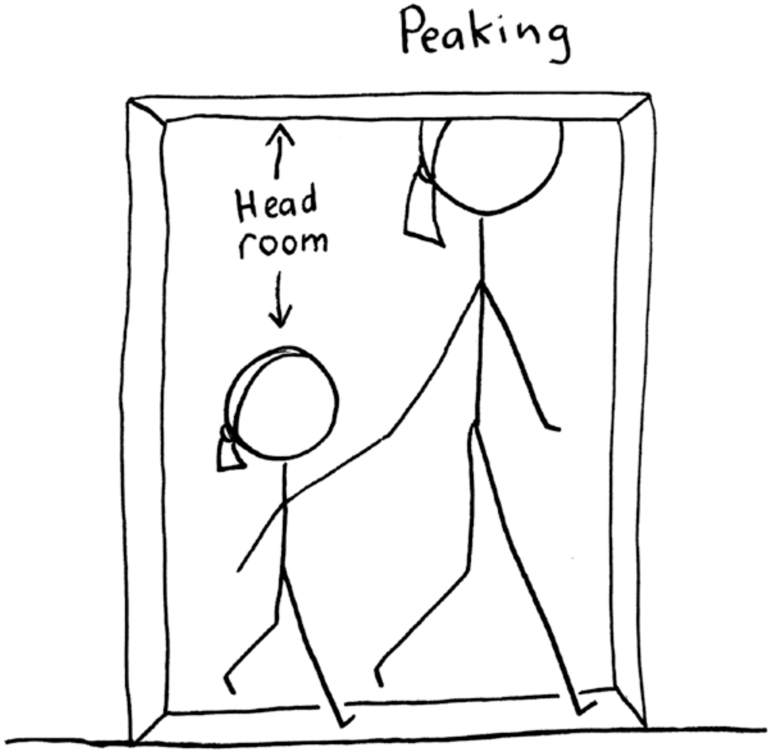

Figure 4-14. *Head room and peaking*

After you have recorded audio that's peaking, you can't really fix it. You can turn down the level on your file in your audio editing program, but the peaks of the audio waveform will remain flat. You're simply never going to get the missing audio data back. It's a bit like what's happening in Figure 4-15.

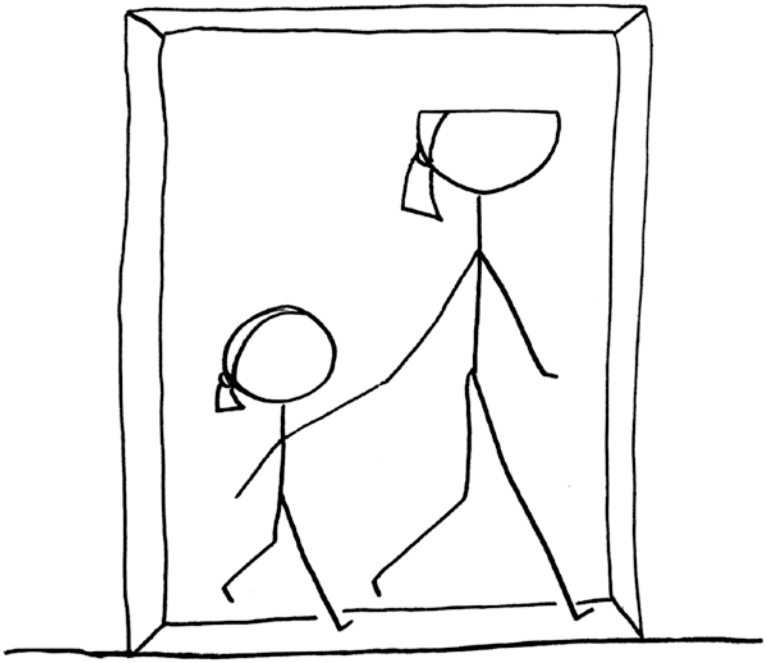

Figure 4-15. *If you record audio that's peaking, you can't really fix it. You can turn down the level later, but you will never recover the missing audio data*

If you have to choose between recording at a slightly low level and peaking, then you should choose the former. The computer is pretty good at filling in the blanks when you turn up a recording that has been made at a low level. It is not that good at repairing audio that has peaked. Generally speaking, you want to find a happy medium between not recording enough audio data and peaking. I will share a standard for this that I have learned in the radio broadcast industry in Australia.

Setting a Level for Recording Human Speech

When setting your level, remember -18dBFS.

I refer here to dBFS, rather than dB. The decibel scale (dB) is used to describe a number of things. dBFS stands for "decibel full scale" and is particular to recording digital audio. I'll simply refer to decibels or dB for the rest of this section.

Whether recording in 16 bits or 24 bits, -18dB gives you a good amount of detail and a good amount of head room. One reason that -18dB is good is because most people laugh about 10dB louder than they speak, so that would leave you about 8dB of head room. If you're recording in 24 bits, you can get away with setting a lower level.

Despite the pages of explanation it took me to get to this point, setting your level is actually quite simple. During your sound check, get the person to speak and adjust the gain control until the level hovers around the point in the meter that says -18 as in Figure 4-16.

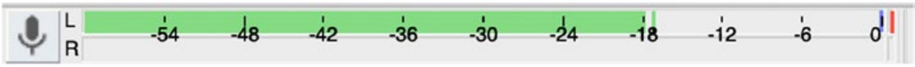

Figure 4-16. *A recording meter (Audacity software)*

Generally, you set your level at the beginning of a recording and leave it. If you do need to change it, do so slowly so that it is not obvious to the listener.

There are a few things to note when reading the Audacity recording meter. These same features will most likely be present in some form on the meters of other software or pieces of equipment. I'll repeat the image of the recording meter so you can follow along.

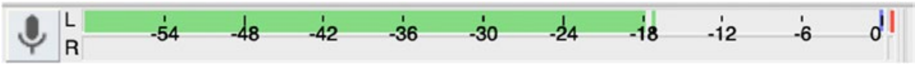

Figure 4-16. *A recording meter (Audacity software)*

The scale on the Audacity recording meter is marked from 0 at the right to -54 at the left. The maximum value is 0, and -54 is somewhere near the minimum. On many meters, the minimum is marked as -∞. When looking at the recording meter on Audacity, the large green area depicts the level that you're at in the moment. The green line to the right of that indicates your recent peak. The blue line indicates the maximum peak for the duration of that recording. The red line indicates if you have peaked at any point in the recording (peaked as in exceeded the maximum value).

Setting a level for recording speech at -18dBFS is technically straightforward, but you'll have to take into account a few elements of human behavior.

Making Allowances for Human Nature

Making a good recording of speech is as much about understanding people as it is about understanding technology. When you are setting the level during a sound check, you should be aware of some predictable human behaviors that affect how loudly a person is speaking.

Even if you have asked your interviewee about their breakfast choices, a person will almost always speak more softly in a sound check than in an interview. So if you set your level at -18dBFS, it might be more like -16dBFS during the interview. This should still allow for enough head room. Another thing to take note of is that when people are speaking English, they tend to start sentences loudly and finish them softly. It's good to have a person say a few sentences to get an idea of the average.

When you're setting your own level, you must resist the temptation to move toward the microphone to get the right level. Instead, you need to put the microphone at the right distance and angle, as discussed earlier, and change the gain control.

When setting your own level, you need to say something that is sufficiently complicated to distract yourself from the fact that you're setting the level, while setting the level, while staying in the right position. This requires some complicated mental gymnastics.

Headphones will affect how loudly a person is speaking. If a person's headphones are too quiet, then they will speak loudly to compensate. If a person's headphones are too loud, then they will speak softly to compensate. Not only can this give you feedback on whether you've set a person's headphones to a comfortable level, you can also use this information to subtly change the volume of a person's speaking voice. For example, if you are having trouble getting a person to speak loudly enough to get a decent amount of signal, then you could turn down their headphones. Generally speaking, you want set the level based on a person's normal speaking volume, but sometimes you have to compensate for less-than-ideal equipment. It's no good asking a person to speak loudly, because as soon as they start thinking about something else such as the interview questions, then they'll revert to their normal speaking volume. Changing the volume of their headphones is much more effective.

Recording human speech requires some understanding of humans. Recording anything else mostly just requires you to think about how loud it is.

Setting a Level for Recording Things That Aren't Human Speech

If you're recording things that aren't human speech, the same considerations apply for setting the level: you want to record enough audio data without peaking.

The level that you choose to record things that aren't human speech should be as loud as you can make it while still having about 10dB of head room. This will depend upon the *dynamic range* of the sound that you're recording, which is the difference between the softest and the loudest parts of the sound. As an example, a dog would have a greater dynamic range than an air conditioner. A dog makes soft panting sounds but also barks loudly, while an air conditioner hums consistently. When setting your level for recording a dog, you will want to set a high enough level to record the panting, but a low enough level so that when the dog barks, the audio doesn't peak. It's safe to record the air conditioner at a high level because it's so consistent.

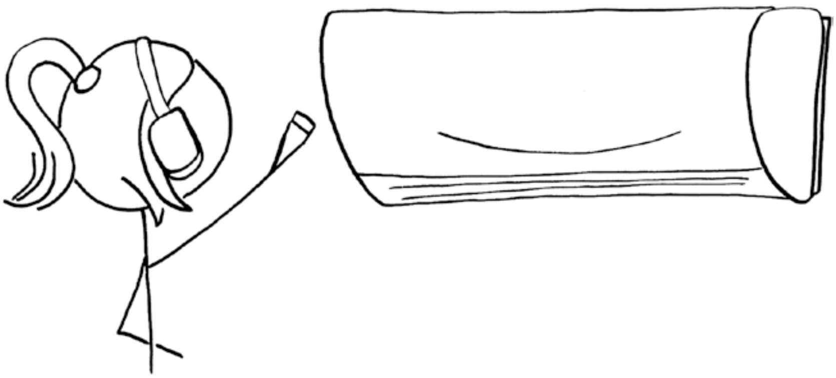

Figure 4-17. *The hum of an air conditioner doesn't have much dynamic range so it's safe to record at a high level*

When you're setting the level of your recording, don't think about how you will want to use that recording. Record it at the best level to capture the sound, and then turn it up or down in the editing process to suit the content. You might use the sound of the air conditioner quietly in the background of your podcast, but you should still get a good recording of it.

Setting levels doesn't have to be complicated, but recording outside can add to the challenge.

Getting a Good Take When Recording Outside

Recording outside often means doing a more difficult job with less. When you're recording in the field, you generally have less equipment and less control over your environment. Most of the time when I've been recording outside, I've had a single microphone, headphones, a portable recorder, and associated paraphernalia. Getting a good take with this setup requires greater concentration.

Often when you're recording outside, you're holding your microphone in your hand instead of using a stand, which means that you might start to hear handling noise. Microphones such as the Sennheiser e945 are designed to minimize handling noise, but a little bit can still come through. To prevent handling noise, try not to tap or bump the microphone or its cable. Try not to move the microphone around too quickly or to shift the microphone around in your hand.

When holding a microphone in your hand, be mindful of which section of the microphone you're holding. The correct place to hold a microphone is on the body. If you cup the grille of the microphone like a fully sick rapper, then you run the risk of disabling its directional capabilities. That is to say, your directional microphone temporarily becomes an omnidirectional microphone and is now picking up a lot more background noise.

Figure 4-18. *Hold the microphone on its body*

When you're making a recording inside, it's best practice to record each person using a different microphone on their own track. This will result in a recording which is much easier to edit. Even when recording outside, it is preferable to record in this way, but it's not always practical. Often when you're recording outside, you're using a single microphone for everyone, so you'll particularly want to make sure that you record everyone at around the same level. The problem is that sound intensity doesn't relate to distance in a linear fashion. A microphone that is a bit further away from a source of sound is picking up a lot less signal. As such, when you're using a single microphone to interview multiple people, it's important to keep the distance from the microphone to the mouth about the same for each person.

The classic mistake is to hold the microphone close to your own mouth but far away from the person that you're interviewing as in Figure 4-19.

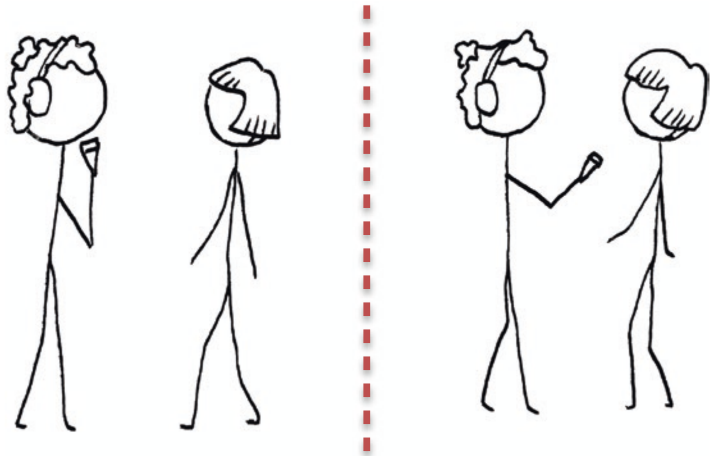

Figure 4-19. *Holding the microphone closer to your own mouth than to the mouth of the person you're interviewing*

If you get this wrong, you end up with a sound file that looks like the one in Figure 4-20 and is a nightmare to edit.

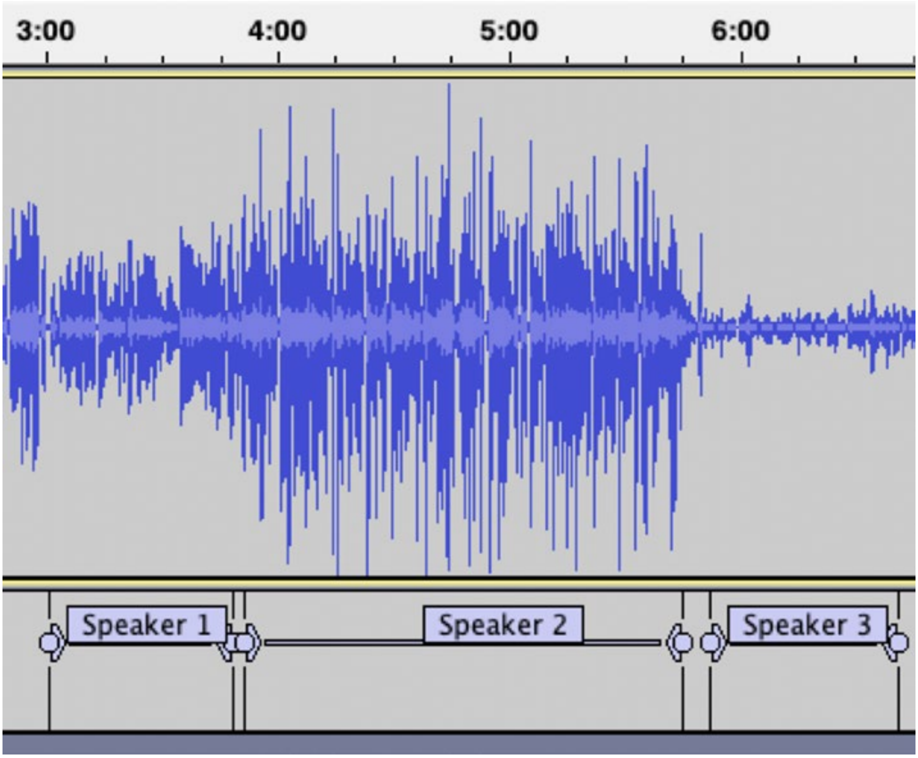

Figure 4-20. *Some dramatically different levels (Audacity software)*

This is four minutes of a file that goes for an hour. In this section, there are three people talking. The audio of the first person is at a good level. The audio of the second person is peaking. The audio of the third person is too quiet. If you left the file like this, then a person who was listening to it would have to repeatedly change the volume on their playback equipment or risk discomfort and maybe even ear damage. But you wouldn't leave a file like this. I will discuss how you would balance these levels in Chapter 8, "Editing" in the subsection "Applying Compression Manually". Better yet, you can prevent recordings like this by being mindful of keeping the distance of the microphone from the mouth about the same for each person.

It's difficult to conduct an interview and also be in charge of audio quality in the best of circumstances, let alone in the field. One way that I have improved my recordings is to rope one of my friends into being the sound person. I set up the equipment myself and sort out the levels with a bit more head room than usual. Then I give my friend a crash course in microphone technique, having them listen to the difference between on-mic and off-mic and impressing upon them the importance of keeping the distance of the microphone from the mouth consistent for each person. Anyone with a set of headphones who's concentrating on audio quality is probably going to do a better job than a person whose mind is on two things at once. One time when I was interviewing a class of schoolchildren, I had one of the kids take on the role of sound person and she did a great job.

Getting a good take in the field can require a little more care, but you can get by with a little help from your friends.

Summary

You probably didn't imagine there could be so many things to take into account when putting a microphone in front of someone's mouth, but here we are. Despite the level of detail in this chapter, most of it comes down to the following. When recording speech, place the microphone about a closed fist away from the speaker's mouth at a 45° angle. The microphone should be pointing at the mouth but out of the way of the breath. Set the gain so you're recording a person's speaking voice at about -18dB. Listen to how it sounds and adjust. Getting the recording right means that your podcast will be easy to edit and it will sound great.

CHAPTER 5

Recording Inside

You might be surprised to hear that I am a champion Bible reader. That's right, when I was growing up, I routinely won the Junior Eisteddfod Bible reading competition in my hometown of Ipswich. As a result of my status as the town's best junior Bible reader, I was invited to read at our local church on Sundays. It was here that I learned my first lessons in acoustics. Churches that predate amplification are purpose-built acoustic spaces. You say something at the front of the church and it rolls down to the back. Then, unfortunately, it comes back. The only way to be understood in this environment is to leave long pauses between your sentences. Priests don't just pause during their sermons so you can reflect on your eternal damnation; there's also acoustics at play. Many people are able to recognize the sound of a voice in a stone church, whether or not they are a champion Bible reader. In fact, most people will be able to hear a lot of information about any space in which they hear a sound. This chapter is primarily about *acoustics*, so we'll be thinking about the qualities of a room that determine how sounds develop in that space. You'll be able to use this information to find good acoustic spaces in which to record your podcast.

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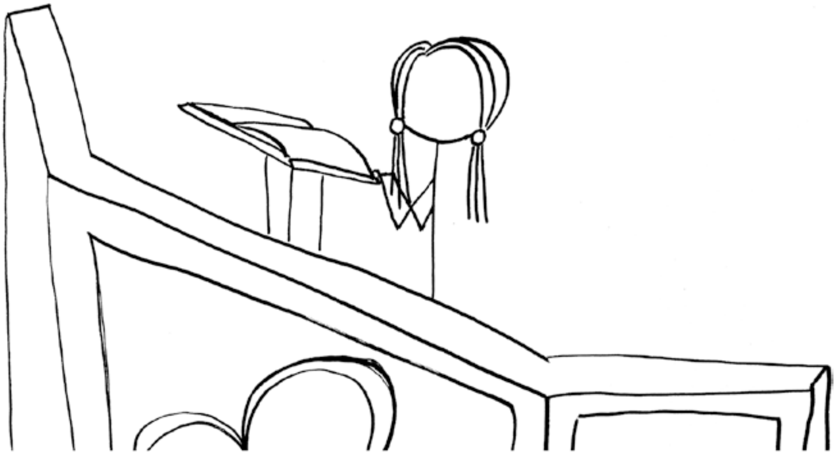

Figure 5-1. *When I was reading for the congregation, I had to consider the acoustics of the church. You will have to consider acoustics when choosing a space in which to record your podcast*

Some or all of your podcast will most likely be recorded inside. This might be in a studio, in your home, or you might be visiting some other indoor space. In Australia, there are a few cost-effective options for recording inside including library facilities and community radio stations. International readers might find similar recording spaces in their local communities. If you choose to record at home, it will require an investment in gear and some knowledge of acoustics.

Recording in a Studio

Before you make an investment in a whole lot of audio gear, you should consider recording in a professional studio.

There are many advantages to recording in a professional recording studio. A professional studio offers high-quality equipment in an acoustically treated space. The cost of hiring a studio ranges from cheap to more expensive, but it is often more economical than investing in equipment for your own home. The audio that you record in a professional studio will generally sound much better than audio that is recorded at home.

You need to be on the lookout for quality. It has come to my attention that some businesses market themselves as podcasting studios, but do not offer the level of service that you would expect from a professional recording studio. There's a lot more to setting up a recording studio than buying some consumer-grade audio equipment and putting it in a room. Luckily, you don't have to be an expert in sound engineering to work out the difference. A recording studio should be quiet, due to the soundproofing. If you can hear outside noises such as the train or the traffic inside the space where you will be recording, then give that studio a miss. Good soundproofing is expensive, but not really flashy, so it's a good sign if the owners of a recording studio have made it a priority.

If you do choose to hire a studio, consider the following affordable options.

Public Libraries and Community Spaces

You might find that a community organization has set up a recording studio in your local area that is available to hire. In Australia, for example, a number of suburban councils and libraries have recording studios available to hire for a very reasonable price. If you can find such a facility, I would highly recommend using it, because this will give you the best value for money. The council or library generally provides training for the use of their equipment. Look for something similar in your local area.

Radio Stations

Consider hiring a studio at your local radio station. There are numerous small radio stations dotted around the world, serving small sections of the community. They often hire out their studios with or without an engineer as an alternative income stream. This is a reasonably affordable option for those making a podcast. You can find your nearest radio station using the **World Radio Map**.

Radio stations have a lot to offer podcasters. The equipment in a radio station will be optimized for recording voice, and the space will be acoustically treated.

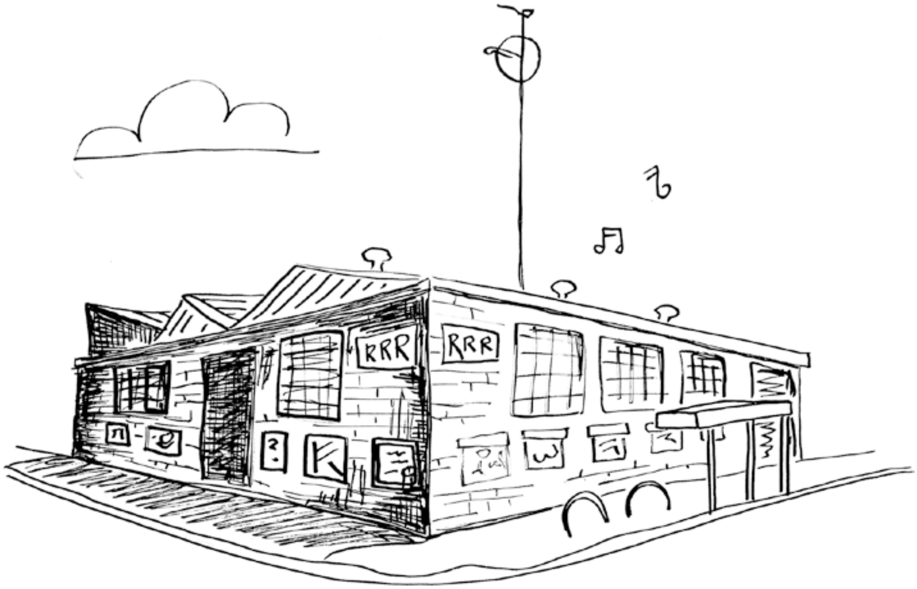

Figure 5-2. *Radio stations are a great place for podcasters to make a recording*

Recording at Home

Recording at home offers a lot of flexibility, but requires a greater investment in equipment. You can make a pretty good recording at home, but it probably won't be as polished as a recording made in a studio.

If you're recording at home, it's worth spending some time finding the best space. The principal issues to consider are background noise and the overall sound of the space. I believe that with the right equipment, you should be able to greatly reduce your recording of background noise. This will enable you to concentrate on finding an area that's comfortable and that sounds good. For further information on equipment, see Chapter 2, "Gear Part 1" and Chapter 3, "Gear Part 2".

Reducing Background Noise

I live in an apartment in the inner suburbs of a capital city. My days begin with the screech of wattle birds. Traffic hum is constant. Conversations drift up through my window. Although I've never met the girl downstairs, I can say with a great deal of certainty that her favorite movie is *The Sound of Music* and that she aspires to be a musician. I can tell you that she will undoubtedly achieve that ambition because she's dedicated to practicing at all hours. Does this situation find me huddled in my closet recording voice-over? Not necessarily.

Figure 5-3. *The musical ambitions of my neighbor don't clash with my desire to make recordings*

When you're recording in a studio, the inside of the room should be quiet due to the soundproofing. When you're recording at home, you'll need a basic understanding of how sound travels through walls to help you minimize the amount of background noise. Imagine that you're walking down the street toward a pub to see your favorite band. The first thing you hear as you approach the pub is the *thump*, *thump*, *thump* of the bass guitar and the kick drum. As you draw closer, you start to hear the muffled roar of electric guitars. When you open the door, it's like the music has suddenly come into focus and you're enveloped in sound. This scenario demonstrates that different sounds travel through the walls of a pub differently. Sounds that are of a low frequency such as bass instruments can easily travel through walls and other heavy objects. Sounds in the middle frequencies such as guitars and voice meet considerable resistance from the wall. Even when they make it through the wall, they sound muffled, because the higher-frequency parts of these sounds are missing. You might hear the singer, but you can't make out the words because you're missing the consonants. This tells you that high-frequency sounds are easily blocked by a wall.

If you stand outside the pub having a chat, you'll notice that every time the door opens, the music is considerably louder and clearer. This is because sound travels a bit like heat – it will use any air gap to escape.

When I'm recording at home, I take these properties of sound into consideration to reduce background noise without too much trouble or expense. I use a microphone that prioritizes the frequencies of human speech and so rejects really low frequencies. As such, I don't worry if small amounts of traffic rumble or other really low-frequency sounds are coming into the room. I concentrate my efforts on dampening sounds in those higher frequencies that really clash with human speech, such as birds. Sounds in these higher frequencies are much easier to deal with. To reduce these sounds, I act as though it's a cold day and I want to keep the room warm. I close the doors, the windows, and the curtains, and I might go so far as to block out the gap underneath the door. In this minimalistic way, I can make a pretty good recording of human speech at home without too much expense or effort.

Once you have taken steps to reduce the recording of outside sounds, you can start to consider how well the inside of your room works for recording.

What's in a Studio?

It's worth looking at the design of a recording studio so you can apply some of these principles to finding a place to record in your own home. When you hire a studio, you're paying for the sound of the room as much as you're paying for the fancy gear and the expertise of the engineers. This is because when you're recording a sound, you're not just recording the direct sound, you're also recording repeats of that sound that have bounced off the surfaces around it. These repeats of the sound are known as *reflections*. Collectively, they are known as a *reverberation*, or *reverb* to sound engineers, who love shortening words almost as much as they love dressing in black denim. A reverberation gives a listener information about the size and shape of the room and the textures of the surfaces within it.

If you were building the hypothetical worst recording studio, it would be a small, bare room with hard, symmetrical walls, ceilings, and floors (see Figure 5-4). Recording in a space like this guarantees poor-quality sound. Think about how you might sound if you were singing in the toilet.

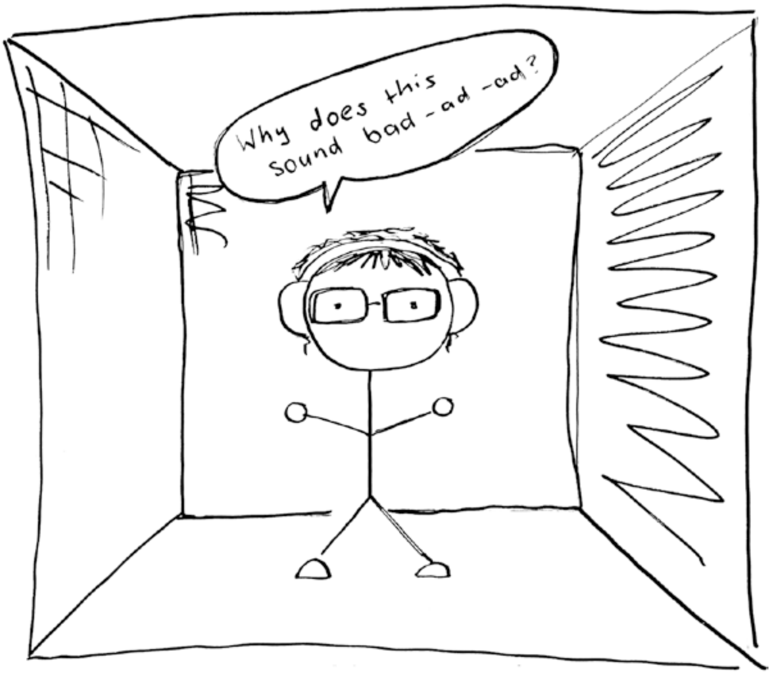

Figure 5-4. *Hypothetical worst studio*

A good studio is basically the opposite of this. Here's what to look for in a studio or in your own home.

Look for Asymmetrical Surfaces to Randomize Reflections

Some people will recommend the toilet as a good place to record at home because it's quiet, but it sounds bad partly because of the echo, but also for another reason.

The symmetrical walls of the toilet are causing sound waves to bounce back on themselves evenly. This causes certain frequencies of the sound to cancel out and others to be accentuated, which sounds unnatural. The more symmetrical a room, the worse this effect is. Audio 5-1 is a recording I made in my toilet to demonstrate.

Audio 5-1 Recording in my toilet

Really good studios have asymmetrical walls, which cause sound to bounce around randomly. If there are symmetrical walls, then they are often broken up with *diffusers* (Figure 5-5). A diffuser is a wall panel with hard uneven surfaces that are designed to disperse sound waves.

Figure 5-5. *A section of a diffuser*

Not only do you want to avoid recording in a room with a lot of flat, symmetrical surfaces, but you also want to mix up the type of surfaces.

Consider the Surfaces of the Walls, Floor, and Ceiling

Recording studios usually contain a mixture of hard surfaces such as wood and soft surfaces such as carpet. The mix of hard and soft surfaces will create a balance between the room sounding too *live* and too *dead*.

A sound wave traveling through a soft surface encounters resistance and so loses energy more quickly than a sound wave bouncing off a hard surface.

If you make a sound in a room with lots of hard surfaces, like the toilet, the sound wave barely encounters any resistance. Too many reflections come back and you hear an echo. You would say that the room is too *live*.

A studio generally contains a certain amount of soft surfaces like carpet to reduce echo and to ameliorate problem frequencies. A studio might have foam panels on the wall or in the corners to attenuate specific frequencies that are a problem in that space.

Recording studios are not completely padded rooms, because that would sound too *dead*. If you make a sound in a *dead* room, then insufficient room reflections come back. Another popular piece of advice for podcasters is that they should record in their closet because, like the toilet, the closet is quiet. If the toilet has too many hard surfaces, then the closet has too many soft surfaces in the form of your clothes. These soft surfaces absorb too much of the high-frequency sounds, making speech sound muffled or dead. Audio 5-2 is a recording I made in my closet to demonstrate how this environment can affect your voice.

Audio 5-2 Recording in my closet

You might sometimes see pictures of people recording in heavily padded vocal booths. These rooms sound dead, but in this case, the sound engineer will add reverb later.

If you want to appreciate the effect of hard surfaces on your voice, then consider how amazing you sound singing in the shower. The hard surfaces of the shower unit reflect your singing back at you making you sound like a superstar. As a shower unit is not a completely symmetrical, enclosed box like a toilet, it doesn't tend to suffer from the same problems. However, it would probably be wrong for recording your podcast as, like the church, it's a pretty distinctive-sounding acoustic space.

It's not just the shape and the surfaces; the size of a room also affects the room reverberation.

Consider the Size of Your Room

Good studios tend to have fairly large rooms, which gives sounds a feeling of space and clarity. In a small room, reflections follow on from the initial sound very quickly, whereas in a large room, there's a longer gap between the initial sound and the first repeat of it. Consider how a musician might sound in a concert hall next to how they might sound in a typical classroom.

The small size of the toilet and closet is another reason to avoid recording in these spaces. A recording made in a small room can sound claustrophobic.

Acoustics is not the only issue you will face when recording in your toilet and closet; there are also practical considerations. These spaces are not at all comfortable, and they're not an appropriate place to bring a guest. While it's good to find somewhere reasonably quiet to record, it doesn't have to be your primary consideration if you're using appropriate equipment.

Figure 5-6. *Recording in a closet is quite impractical*

The size, shape, and textures of a room affect the way that sound waves reverberate. You want some reflections but not too many. You want them to come back to you but not too quickly. You want them dispersed randomly. Now that you know what to look for, it should not be difficult to find a suitable place to record at home.

Finding a Place to Record at Home

Think back to the idea of the hypothetical worst studio – the small room with hard, symmetrical walls, ceiling, and floors. When recording at home, try to find a space that is the opposite of that.

Look for a fairly large room. If you have an asymmetrical room available, this might be a good choice. If not, then keep in mind that furniture helps to break up symmetrical surfaces and so randomize the way that sound bounces around. Smaller objects will tend to randomize the reverberation further. So a bookshelf is good, and a bookshelf full of books is better. It's good to record in a room that has a variety of surfaces, such as carpet, plaster, wood, and windows to find a balance between a live and dead sounding space.

You probably don't have soundproofing in your home, but look for a quiet room. Sounds that are in a really low frequency range such as traffic rumble will be less of a problem than sounds that are in the same frequency range as human speech, such as birds.

When I record at home, I record in my living room, which is not acoustically treated. Listen to Audio 5-3 to hear how my voice sounds in my living room.

Audio 5-3 Recording in my living room

My living room is a fairly large room. It is a symmetrical box, but it is full of furniture and other objects. You will know from your own experience that an empty room has a different sound to a room with furniture and belongings in it. Some might say that my living room is full of clutter, but I think of it as acoustic treatment.

My living room contains some hard surfaces such as plaster walls and windows and some soft surfaces such as carpeting, so it is a good mixture between live and dead.

Not only does my living room sound good, it's also a comfortable place to work and a suitable location to conduct an interview.

Unfortunately, my living room isn't particularly quiet. Usually it's fine, but sometimes when the noise becomes too loud, I use my bedroom instead. My bedroom has smaller windows which let in less noise.

Now that you know what to look for, you can find the ideal area in your home. Furthermore, you can use this knowledge to find a place to make a field recording inside any building.

Listening to the Space

Understanding acoustic principles will help you narrow down a few likely places to record, but the real test is how these places sound. Once you have found a likely space, set up your equipment and have a listen. Pay attention to background noises and listen to the way your voice sounds in the space.

You might think that a room is quiet, but when you listen through recording equipment, tiny noises can become apparent. When I put on my headphones and listen through my setup, I notice that the traffic rumble reduces, but the clock starts yelling at me, "TICK, TICK, TICK." My microphone is designed to accentuate vocals, which is why it also accentuates the ticking of my clock. The ticking sound is very much like a consonant – a "T" or a "K." The solution is to move my clock to another room when I'm recording.

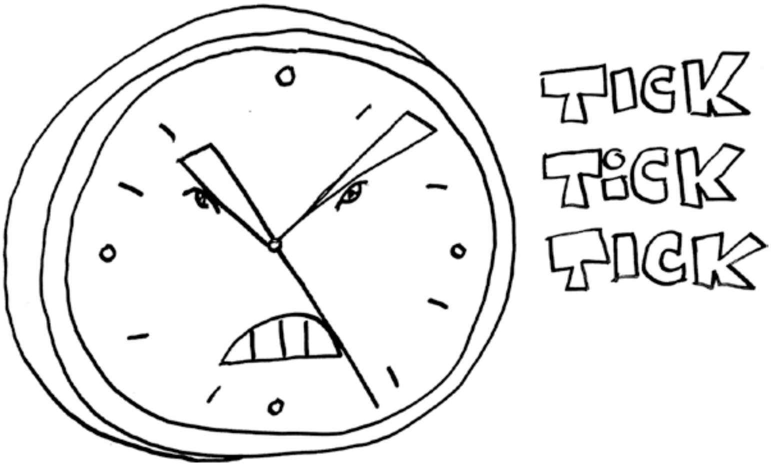

Figure 5-7. *You might find that your recording equipment accentuates little sounds such as the ticking of a clock*

When you've found an area that sounds good, make a recording and listen to how your voice sounds in the space. If you're comparing a few spaces, you should note that your voice will sound louder or softer in different rooms. Make sure to adjust the volume of your recordings to make a fair comparison.

Using your ears and some basic acoustic principles, you can find the best place in your home to record a podcast.

Summary

A professional recording studio will give you the highest-quality recording and could be cheaper than investing in your own equipment.

If you're recording from home, you should consider the acoustics of the room that you're recording in. When looking for a space to record in your home, you should find a medium to large room with multiple types of surfaces such as carpet and wood and with lots of furniture and other objects within it. Imagine the worst hypothetical studio – the small, empty cube with bare walls – then find the opposite of that.

A room with less background noise is better, but you don't have to record in your closet. You should consider your comfort and that you might want to invite other people around to conduct an interview. Use equipment that doesn't pick up too much background noise. Dampen outside noises the same way that you would keep heat in a room.

Set up your equipment, make a recording, and listen to how you sound in a few different places.

Your ears and your knowledge of acoustics will help you find suitable places to record indoors.

CHAPTER 6

Recording Outside

Recording outside is a challenge, but the challenge is often rewarded with interesting content, a greater diversity of voices, and audio that takes your listener someplace else.

I have this friend Andy who is deeply committed to his anarchist and Christian principles and to living life on his own terms. That has meant a lot of things: living in a house where anyone is welcome to stay, living for a year without money, chaining himself to various things. I admire that he has the courage to walk the talk.

He's a musician and is most famous for his song *Don't Kill People* which gets a sing-along at the right parties.

C. M. Green, *The Podcaster's Audio Handbook*,
https://doi.org/10.1007/978-1-4842-7361-6_6

Figure 6-1. *My friend Andy*

In 2016, Andy was part of a group who walked onto Pine Gap, the secretive military base in the Australian desert, and sang a lament. Pine Gap is a satellite surveillance base that plays a significant part in the United States' military effort. The people who run Pine Gap were not impressed with having activists walk onto their base. Perhaps they didn't like the singing. Whatever it was, Andy and his friends were in Serious Trouble.

The next year, they went on trial in the Northern Territory Supreme Court. I went up to the trial to support Andy, and I ended up doing a story about it. In terms of telling the story, I could have stayed at home and interviewed Andy remotely, but it would not have been the same. He probably would have given me information about the court proceedings and the political aims of his group, but what I witnessed went far beyond that.

During the trial, I stayed with Andy and his friends and got to see who they were and how they lived. Sitting in the blazing sun outside the courthouse, I spoke to peace activists from all walks of life. I learned what drives people to take up this issue with such passion and how the global military-industrial complex permeates so many other matters.

In the large, air-conditioned courtroom, I watched the spectacle of a group of scruffy activists defending themselves against wall-to-wall government lawyers and witnesses. A turning point in the case was when one of the activists cried as he spoke about the horror of the lethal drone strikes that were being directed from Pine Gap. This moment made a complex issue feel as simple as Andy's song: *Don't Kill People*.

Recording in the field threw up some challenges. I interviewed an academic on a busy road and a former senator in a noisy park on a windy day. I plugged into the PA system at a public meeting. I had to keep my equipment safe and stay organized. The different recordings offered their own technical challenges and also gave the story some of the flavor of the trial.

Recording outside can mean finding something unpredictable, leaving your comfort zone, discovering a different perspective, or telling a story in vivid detail. Not everyone has the resources or inclination to leave their locality to find a story, but there is sure to be something interesting happening in your area. It's up to you to find it.

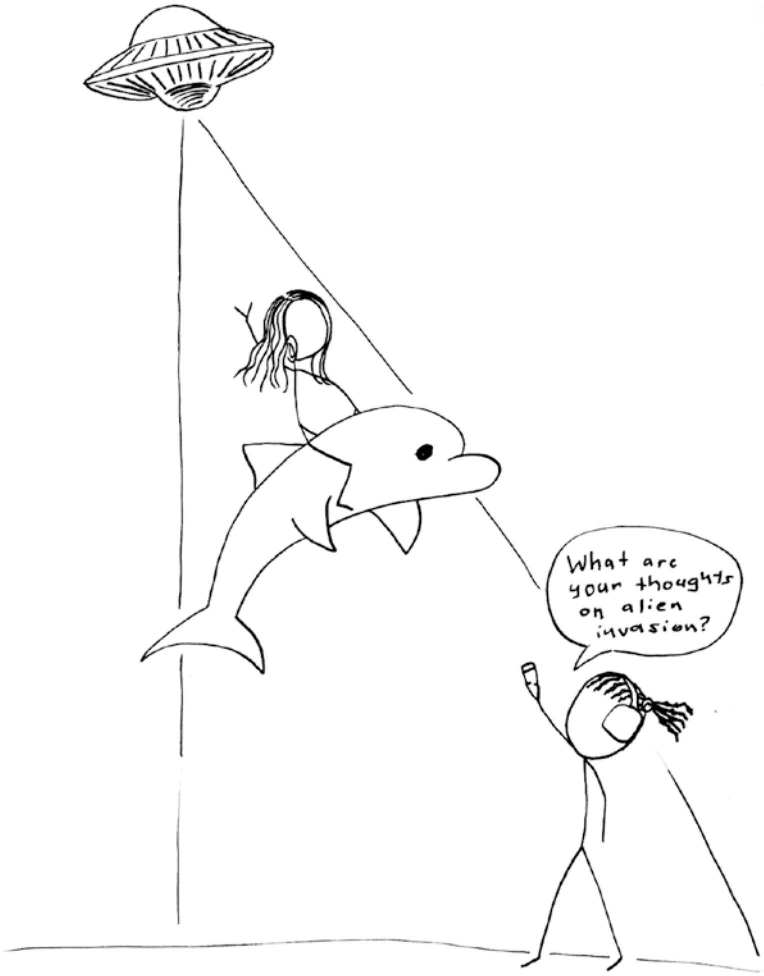

Figure 6-2. *A lot of interesting things happen outside*

Recording outside can also be a great way to bring your fiction podcast to life. All of the techniques I discuss in this chapter can be employed to create a world that your listener will become immersed in.

In this chapter, I'm going to show you the settings you'll need to use on a portable recorder; how to work with background noise, sound effects, and the weather; and how to record on location at events. I want to equip you with the knowledge to record in different situations and show you that technical choices can aid in creative storytelling.

Be Prepared

Recording outside can be unpredictable, so you need to be prepared.

Before you head out into the field, familiarize yourself with your equipment. Assemble and test the equipment the day before your recording to make sure everything is there and it is working. Charge the batteries on your equipment, and charge your own batteries with a good night's sleep.

An important part of preparation is packing your bag. Consider the situation you will be entering, and pack only what you need.

When recording outside, I recommend using a portable recorder, headphones, and a dynamic microphone. For more information, see Chapter 2, "Gear Part 1" and Chapter 3, "Gear Part 2 - Microphones". If you are recording in a building, you can bring a more complicated setup similar to what you would use at home.

When I'm recording outside, I pack all of my gear into a waterproof plastic food storage container. I have found that Lock & Lock containers are reliably waterproof. Some people use specialist padded cases that offer protection against shock, but the compromise is the size and weight. Consider your own situation.

I pack the minimum, but I do bring one more balanced cable with XLR connectors than I will need, as these can be unreliable.

If I'm going to be recording at an event that has a *sound reinforcement system*, then I'll bring a balanced cable with TRS connectors and a cable with both a TRS connector and an RCA connector. A sound reinforcement system is an audio setup that involves microphones, speakers, etc., and is used to amplify sound and distribute it to an audience. Using the cables mentioned earlier, I can plug into the sound reinforcement system and record a direct feed of the event's proceedings. I'll discuss this in more detail later. I bring both cables because I don't usually know which one I'll need.

I bring spare batteries, and if I anticipate doing a lot of recording, I will also bring a spare SD card. Along with my usual mic sock, I pack a wind sock, even if it's not windy. I bring a pen and a notepad, so I can write down important information.

And at risk of sounding like your mother, when packing your bag you should consider your own needs: lunch, water, and practical clothing for the weather. It would be a shame to miss out on the action because you're cold or hungry.

There are many reasons you may want to record outside, so it's hard to be more specific in this advice, but time spent researching and preparing is usually time well spent. If you're recording at an event, liaise with the event coordinators to get their permission and information about the location and proceedings. If you're gathering news, then you should understand the main elements of the story and the main players, and so on.

Planning your outing will help you record the best audio and have the most enjoyable experience.

Settings on a Portable Recorder

Portable recorders offer wonderful flexibility for recording outside the home. Understanding the settings on your portable recorder is a key part of using it well. In this chapter, I am going to use the Zoom H5 (Figure 6-3) as an example, but other portable recorders should have similar features. I have put the directions for the Zoom H5 in boxes, so that they are easy to skip over if they are irrelevant to you. To learn more about the Zoom H5 portable recorder, go to Chapter 2, "Gear Part 1."

Figure 6-3. *A Zoom H5 portable recorder (photo courtesy of Zoom)*

Before using your portable recorder, go into the menu and change the file format and settings for recording as discussed in Chapter 1, “File Formats and Settings.” You’ll want to be recording wav files that are 24 bits and 44.1 kHz.

If you’re using the Zoom H5, go to the menu. Choose REC ➤ REC FORMAT. Select WAV44.1kHz/24bit.

Set the date on your portable recorder to help keep track of what you have recorded.

On the Zoom H5, go to the menu SYSTEM ➤ DATE/TIME and set the date and time.

Consider what you're recording and how you plan to record it when setting the channel count on your portable recorder.

You can record in stereo using the inbuilt microphones of a portable recorder. This is useful for recording atmos.

If you're using external microphone(s) with your portable recorder, you should record each channel separately, or in mono. If you've plugged in two microphones and are recording in stereo, then you are creating one file. It's more useful to have two mono files because it's easier to edit different voices separately, although it is easy enough to rectify this in the editing process.

The Zoom H5 records in stereo as default, but if you want to record mono files or a combination of mono and stereo files, then you should choose *multi-file mode*.

Go into the menu and choose REC MODE ➤ MULTI FILE. I recommend leaving the recorder on this setting all the time.

Navigate back to the main screen, and you can now select the microphones that you wish to use. To select the inbuilt stereo microphones, engage the buttons on the face of the recorder labeled "L" and "R." To select external microphones, or anything else coming in through the channels, engage the button(s) on the face of the recorder labeled "1" and/or "2."

Each of these channels has their own gain control. The gain controls for channels 1 and 2 are underneath the main screen. The gain control for the inbuilt microphones is situated just underneath those microphones.

If there's a high pass filter (also known as a HPF and also known as a low cut filter) available on your recorder, then turn that on for recording human speech. This filters out really low-frequency sounds such as traffic rumble or air conditioning. Depending on the portable recorder, the high pass filter will roll off frequencies lower than about 80 to 100 Hz. You will be impressed by how much louder and clearer speech sounds when your high pass filter is engaged.

To turn on the high pass filter on the Zoom H5, go to menu ➤ IN/OUT ➤ LO CUT, and choose which inputs you want to apply the high pass filter to. There are three options for your high pass filter, and I recommend 80 Hz.

If you're using a microphone that requires external power, engage phantom power in that channel. This could also be marked as +48V, +48, 48V, or something similar.

If you're using the Zoom H5, you can turn on phantom power by going to the menu and choosing IN/OUT ➤ In1/2 PHANTOM ➤ ON/OFF and then choosing your input channels.

Some portable recorders require you to press the record button twice to actually record. The first time that you press the record button, you're only monitoring what's coming through the microphones or whatever else is plugged into your portable recorder. This "feature" is the cause of many missed opportunities and great heartbreak. The Zoom H5 starts monitoring when you select your channel and so only requires you to press the record button once to make a recording. If you're using a different portable recorder, then find out exactly how the record function works before you go out into the field. When you're recording outside, make sure that you're actually recording by checking whether the time counter is ticking over.

Familiarize yourself with the settings on your portable recorder so you can use it like a pro.

Background Noise

A lot of the time when you're recording outside, you're going to be recording someone speaking. When recording speech, it's desirable to get a clear recording without too much background noise. There are numerous strategies to minimize the recording of background noise, and I am going to go through five in order of easy to difficult.

When making a recording outside, it is not feasible to completely eliminate background noise. You can think of the background noise as setting a scene. If you really like the ambient sound and want to include it in your recording, you can record that as well and layer it with the speech in the editing stage. I'll discuss that more later.

Always wear headphones when recording. In this context, the headphones will give you valuable information as to how much background noise you are picking up.

Strategy 1: Find the Right Location

Before you conduct an interview, find a space that sounds good through your recording equipment. Once you step outside, there are many factors influencing background noise and the sound of different spaces, so there are no hard and fast rules as to which spaces will sound good. A slight change in topography can mean that two areas only a short distance away from each other can sound quite different. First, listen with your ears; then, listen through your equipment. Your equipment will pick up sound in a different way to your ears, so it's an essential check. Once you've finalized a location, find your interviewee and lead them to that space.

Strategy 2: Plug an External Microphone into Your Portable Recorder

Plugging an external microphone into your portable recorder will help you to minimize background noise because you can use it to focus on a particular sound. For more information on the type of microphone that you should use, check out Chapter 3, "Gear Part 2 – Microphones."

In Chapter 2, "Gear Part 1," I shared some recordings to demonstrate the difference between recording speech in a noisy environment with the inbuilt microphones of a portable recorder and with an external microphone. I will repeat those recordings here so you can listen to how much clearer the speech sounds in the latter recording.

Audio 6-1 Recording with inbuilt microphones of Zoom H5 at a construction site

Audio 6-2 Recording made with a portable recorder and an external microphone at a construction site

If you're using a microphone, plug it into an XLR input (Figures 6-4 and 6-5) of your portable recorder with a balanced cable (Figure 6-6). Don't forget to use a windjammer microphone sock (Figure 6-7) on your microphone. Select the appropriate input channel(s) on your portable recorder.

Figure 6-4. *An XLR input (Piotr Piatrouski/Shutterstock.com)*

Figure 6-5. *The inputs on a Zoom H5 portable recorder are combined XLR and quarter inch sockets (photo courtesy of Zoom)*

Figure 6-6. *A balanced cable with XLR connectors (photo by Karoline Morwitzer)*

Figure 6-7. *A windjammer mic sock (photo by Karoline Morwitzer)*

To test if you have selected only the external microphone and not one of the inbuilt microphones, lightly scratch the grille of each of these microphones.

Using an external microphone will greatly enhance your ability to record speech clearly in a noisy environment, but this might not be enough on its own.

Strategy 3: Move the Microphone Closer to the Mouth of the Person Talking

A simple strategy for recording in a noisy environment is to turn down the gain and move the microphone closer to the mouth of the person talking. Audio 6-3 is a recording I made to demonstrate the effectiveness of this technique.

Audio 6-3 Moving the mic closer to my mouth to reduce background noise

Holding the microphone closer to the mouth may result in increased recording of mouth noise, but hopefully, the mouth noise will be disguised by the background noise. If you're recording in this way, be careful to avoid sibilance and popping.

This strategy can easily be combined with the next strategy which is pointing the microphone away from the source of background noise.

Strategy 4: Understand the Polar Pattern of Your Microphone

To make a clear recording in a noisy environment, it's useful to understand the ways in which your microphone has been designed to reject unwanted noise. This knowledge will help you point the microphone in the best direction.

In Chapter 3, "Gear Part 2 – Microphones" (section "Polar Patterns and How They Help Reject Background Noise"), I briefly explained that microphones have polar patterns. That is, each microphone is designed to pick up sound from certain directions and reject it from others. If you haven't read that section, I recommend doing so now. In this section, I want to explore the polar patterns of microphones in more detail and discuss how to take advantage of the polar pattern of your microphone while making a field recording.

Polar patterns are only approximate. The ability of a microphone to pick up sounds from certain directions and reject it from others depends on the frequency of the sound in question. High- to middle-frequency sounds including talking will be rejected quite effectively, and low-frequency sounds such as traffic rumble will not be rejected that well.

You can hear how well the Sennheiser e945 microphone both accepts and rejects sound in the frequency range of speech in the audio clip where I demonstrate being on-mic and off-mic.

Audio 6-4 On-mic and off-mic

The ability of a microphone to prioritize sound within the range of human speech from the front and reject sound within the range of human speech from other directions is useful if you are interviewing someone at a busy event. The sound of the crowd will be recorded at a much lower level than your interviewee.

While directional microphones are less effective at rejecting lower-frequency sounds like traffic rumble from any direction, these sounds will not obscure a person's speech as much.

There are a few strategies for taking advantage of the polar pattern of your microphone. If you are recording one person speaking, then you can face them toward a source of noise, so that their microphone is pointing away from the source of noise. I've made a recording (Audio 6-5) so you can hear the result. In this test, I'm facing the source of noise (which is a hair dryer), so when I'm speaking, the microphone is pointing away from the hair dryer. My friend Farida has her back to the hair dryer, so when she's speaking, the microphone is facing toward the hair dryer. You can hear that when I point the microphone toward Farida, the noise of the hair dryer gets slightly louder, particularly in the mid-range frequencies that compete most with human speech. The difference in the background noise is not huge, but my speech is clearer than Farida's, even though she's a bit louder.

Audio 6-5 Recording speech with one person facing a source of noise and one person facing away from the source of noise

When you're recording a single person, it's easy to face them in the direction that noise is coming from. Conducting an interview is slightly more involved because eye contact is critical to holding an effective conversation. You can't face both people toward the source of noise, but you can place them parallel to the source of noise (see Figure 6-8). You will still be rejecting some background noise from this position, and more importantly the noise in the recording will be fairly even between both parties.

***Figure 6-8.** Position yourself and the interviewee parallel to the source of noise*

I recorded Professor Richard Tanter in this manner on the main road of Alice Springs during the Pine Gap trial (Audio 6-6). Ordinarily, I wouldn't record on a busy road, but this was an extraordinary circumstance. Professor Tanter is a world expert on peace, security, and environmental issues and a joint recipient of the Nobel Peace Prize. He had acted as an expert witness in the trial and was running off to catch a plane, but had kindly agreed to a quick interview. I thought despite the busy road, my listeners would want to hear what he had to say. While the traffic is evident in this recording, it stays around the same volume and therefore doesn't take attention away from the interview.

Audio 6-6 Recording parallel to a busy road

An alternative to this method is to face the interviewee toward the source of noise. Their side of the interview will have the least background noise, and your side will have the most. In the editing stage, you can remove your side of the interview and replace with voice-over along the lines of "I asked Zara to give her view on oranges" or "Zara further elaborated on her research into oranges." In this case, you'll have to take extra care not to talk over your interviewee.

Understanding polar patterns will help you skillfully point your microphone in the right direction. But what do you do when there's unwanted noise in every direction?

Strategy 5: Choose the Right Kind of Background Noise

When recording speech outside, consider the nature of the unwanted noise, because some types of noise are easier to deal with in the editing stage than others.

If you're in the position where you have to choose between a number of noisy spaces, you'll find that it's much easier to edit out rumble and other low-frequency noises than to edit out higher-frequency noises. The closer in frequency that the unwanted noise is to speech, the harder it will be to improve the intelligibility of your recording in the editing stage.

Similarly, it will be easier to improve the intelligibility of a recording of speech that has a repetitive noise in the background, such as air conditioner drone, rather than background noise that is random and organic such as parrots.

Figure 6-9. *It's hard to improve the intelligibility of a recording of human speech that has a lot of parrots in the background, because the sound of parrots is random and organic and roughly in the same frequency range as human speech*

Sound engineers have spent a lot of time thinking about how to make a clear recording of speech without too much background noise. You can minimize the recording of background noise by using an external microphone, with your choice of microphone, with the positioning of your microphone, and with your choice of location. Use a combination of these strategies to make the best recording of speech outside. However, not all background noise is unwanted. Now that I've discussed how to reduce the recording of background noise, I'd like you to consider embracing it.

Using Atmos to Tell a Story

Ambient sound can be distracting, or it can set a scene. When it's unwanted, it's considered "background noise," and when it's desirable, it's considered "atmos." You can make use of atmos to enhance your storytelling.

In early 2012, I traveled to a protest against a coal seam gas project in Kerry. Kerry is a small farming community in the Scenic Rim in rural Queensland. The coal seam gas project was in the exploratory drilling stage, and the locals were worried about the effect it would have on the natural environment if it were to proceed. The interviews I conducted contained two types of atmos. In some interviews, you could hear the sounds of the beautiful rural setting, and in other interviews, you could hear the oppressive drone of the drill. These opposing soundscapes illustrate the theme of the story: farmers who didn't want their natural environment damaged by mining. I've summarized the story with two short grabs (Audio 6-7) so that you can appreciate the effect.

Audio 6-7 Using atmos to tell a story

While you might want to use the ambient sound to set a scene, you don't want it to overwhelm your recording. The easy solution is to record the atmos separately to the interview and mix them together in your edit.

To record atmos, you can use the inbuilt microphones of your portable recorder. Previously, I've made a point of saying that these microphones pick up too much background noise, but this is their time to shine. If you're really keen on recording atmos, you can make use of different sorts of microphones suited to this purpose, but the inbuilt microphones of a portable recorder should be of a high enough standard for most podcasts.

If you have the option of omnidirectional microphones on your portable recorder, choose these. You will remember that omnidirectional microphones pick up sound from every direction. These microphones will probably look like two tiny holes somewhere on the body of your recorder, and they will be marked with something like "OMNI" (see Figure 6-10).

Figure 6-10. *The left omni mic on the TASCAM DR-100 mkii*

Locate the holes before you start recording so that you don't accidentally cover them up with your hand. Even though these microphones pick up sound from every direction, your body will be blocking the sound somewhat. Move around until you find the place where the recording of the atmos sounds best through your headphones. If your portable recorder doesn't have omnidirectional microphones, then the other inbuilt microphones will be good for the job.

The Zoom H5 doesn't have omnidirectional microphones, but the inbuilt directional microphones on the end of the unit are good for recording atmos. To select these, engage the buttons on the face of the unit marked "L" and "R." These microphones have their own gain knob. The Zoom H5 inbuilt microphones tend to pick up any amount of breeze, so cover them with a windjammer microphone sock.

Before you start recording atmos, think about how you might want to use it, so you can decide how long the recording should be. Speak into the microphones at the end of the recording to make a note of what you have recorded and why.

When making a field recording of a person talking, you will want to get a clear recording of speech with just the right amount of atmos. The best way to reach this balance is to record the speech and the atmos separately and then mix the two recordings together in the editing phase.

Creating a Balance Between Speech and Atmos

Having separate recordings of speech and atmos enables you to do a number of things in the editing stage. You can create a balance between the atmos and the interview. You can fade the atmos in, out, and around the interview. You can keep running the atmos under the voice-over. You can apply separate processing to the atmos and the interview to bring out the best in both.

Interviews that are recorded outside will have a certain amount of atmos present, even if they are recorded with an external microphone. This is why it is nifty that you can simultaneously record using your external microphone channel(s) and your inbuilt microphones on many modern portable recorders, including the Zoom H5.

To simultaneously record using the inbuilt microphones and external microphones on the Zoom H5, engage the buttons on the face marked "L" and "R," as well as "1" and/or "2." If this isn't working, check that you're in multi-file mode. Go into the menu and choose REC MODE ➤ MULTI FILE.

I have made a field recording in front of a small waterfall at my local canal so you can hear the result of this recording method.

Audio 6-8 Simultaneous recording of atmos and speech

When making a simultaneous recording, you will need to do separate sound checks for each source. Before pressing record, listen through your headphones to the different microphones separately, and set the levels separately.

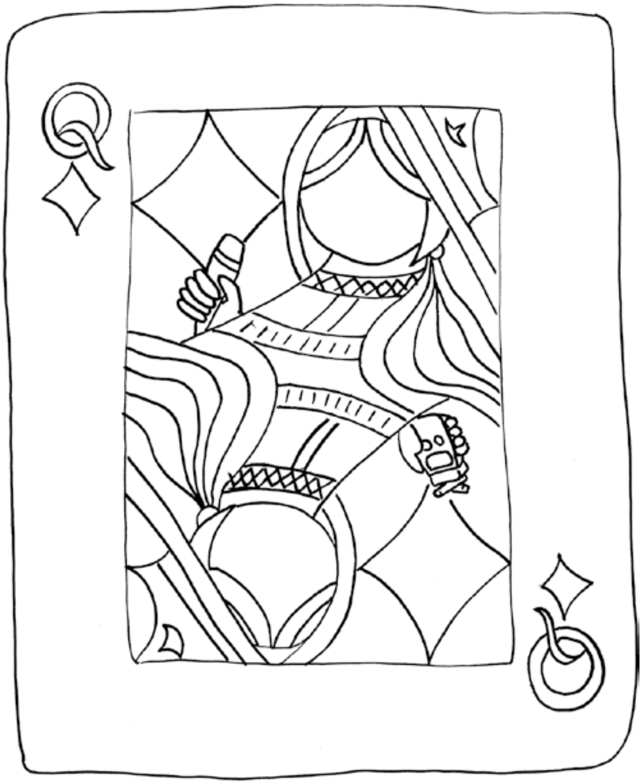

Figure 6-11. *Create the perfect balance by recording speech and atmos on their own channels*

It might be advantageous to record your atmos and your interview in slightly different locations. It could be good to record the atmos where it's clear and loud and the interview where there's a bit less background noise. Recording speech and atmos at separate times is also easier because you're giving each recording your full attention. This is an example of a recording I made at that same waterfall where I combined separate takes of atmos and the speech.

Audio 6-9 Combining separate takes of atmos and speech

It's not noticeable that I have blended together two recordings made at different times that both feature the sound of the waterfall because this sound is regular and repetitive. You'd probably even get away with it if your atmos was something like birds – the two recordings mixed together could just sound like more birds. This technique would not be as effective in a situation where the atmos is more discernible, such as a recording made at an event where music is playing in the background. In that case, you would have to make simultaneous recordings.

Atmos is a great tool for bringing listeners into your world, and sound effects will further aid in the storytelling process.

Recording Sound Effects

If you're recording in the field, it's good practice to look around and see if there are any sound effects you might want to use in your story.

One time I found myself in a shed in Cootamundra watching people shear sheep. I got talking to the wool grader, and it turned out she was a passionate environmentalist. She told me about a project where farmers were breeding environmentally friendly sheep, and she put me in contact with someone to interview. I wanted to get a recording of the sheep to use in my story so I went out to the paddock, stood in front of the sheep, and stuck out my portable recorder. Unfortunately, the sheep didn't know what I wanted. I waited for them to make a sound, and they stared at me like a room full of unimpressed teenagers. I kept waiting and they kept staring. I can see on my recording that this went on for a full awkward minute and a half. Eventually, I realized that to get the sheep to talk to me, I would have to talk to them. That brings us to Audio 6-10 where you can hear my sheep impressions.

Audio 6-10 Corey's sheep impressions

Success! So there's your hot tip for recording sheep. You might have noticed that the clip contained a lot of background noise: the insects in the field and the sound of a pop song blaring out from the shearing shed. This is because I recorded using the inbuilt microphones of my portable recorder, which brings me to my next point. When you're in the field and you're recording a single source sound effect such as an animal, dripping water, or the sound of a bouncing ball, consider using your external microphone. I recommend using the same microphone that you're using for interviews. If you really get into recording sound effects, there's a range of different microphones that are specifically intended for this task, but this setup should be fine for most podcasts.

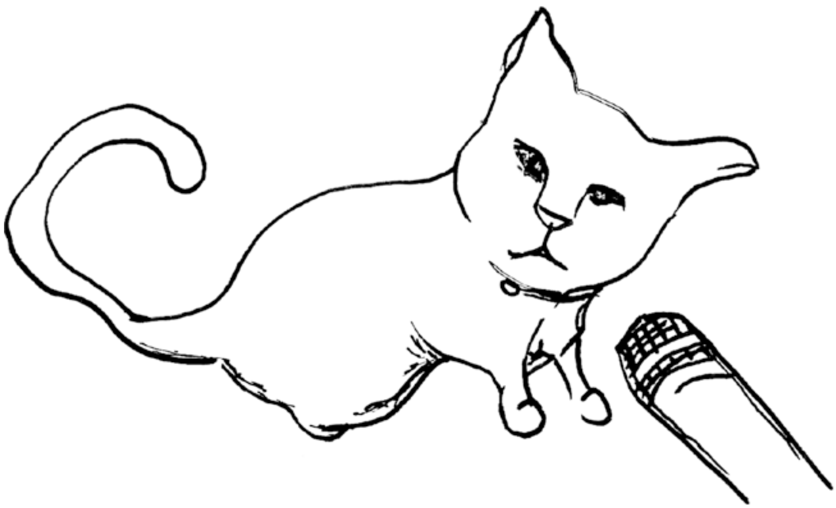

Figure 6-12. *When recording a single source sound effect such as a cat, consider using your external microphone*

I have recorded the sound of a drain with an external microphone (Audio 6-11) and with the inbuilt microphones of the portable recorder (Audio 6-12) so you can appreciate the difference that microphone choice will make.

Audio 6-11 Recording of drain using external microphone

Audio 6-12 Recording of drain using the inbuilt microphones of portable recorder

The recording of the drain made with the external microphone is just that. The recording of the drain made with the inbuilt microphones also includes the sound of the traffic going by. The latter recording is also in stereo, so it conjures up the feeling of a physical space. Your choice here is once again a matter of creative expression – which option works better with your story?

Consider the High Pass Filter When Recording Atmos or Sound Effects

It's good practice to use the high pass filter for recording speech. However, when recording atmos or sound effects, you should tailor your choice to the circumstances.

To toggle the high pass filter on the Zoom H5, go to the menu ➤ IN/OUT ➤ LO CUT, and choose the inputs that you want to apply the high pass filter to. I recommend choosing 80 Hz for recording speech.

The high pass filter removes very low-frequency sounds from recordings. If you want to record a sound with a deep rumble such as a train or a plane or a drain, you will need to turn the high pass filter off. Elephants create sounds as low as 1 Hz to communicate over distances as far as 10 kilometers,[1] so turn off the high pass filter to record an elephant. The high pass filter should be off to record music so that you capture the full range of the instruments, with the exception of instruments that are very high pitched such as flutes.

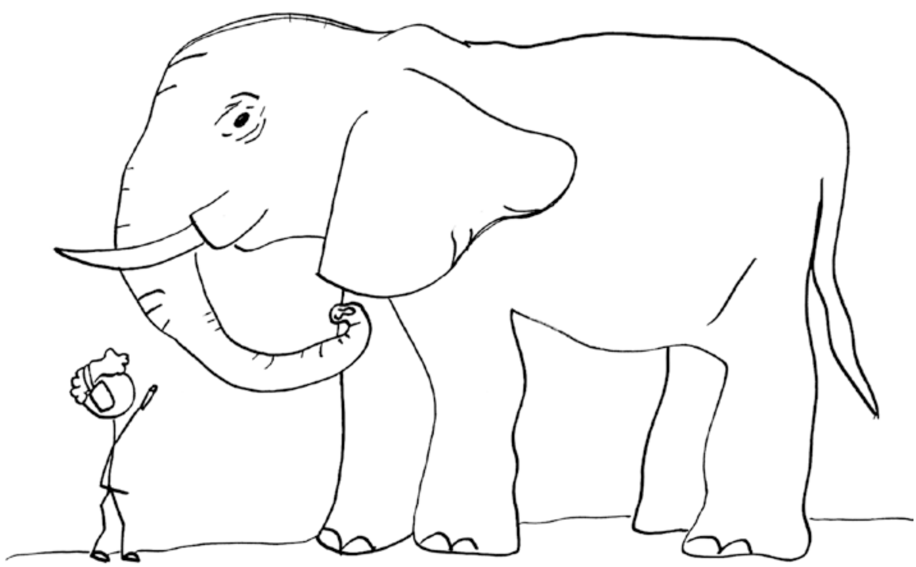

Figure 6-13. *Turn off the high pass filter to record low-frequency sounds, such as an elephant*

Storytelling can influence your choice on whether to use a high pass filter. You might be recording birds in your street. If you wish to reduce the sound of distant traffic in your recording, then your high pass filter should be on. However, the focus of your story might be the city encroaching on

[1] St. Fleur, N., 2020. *An Elephant's Silent Call.* [online] Science | AAAS. Available at: www.sciencemag.org/news/2012/08/elephants-silent-call#:~:text=With%20their%20trumpet%2Dlike%20calls,as%20large%20as%2010%20kilometers. [Accessed 7 September 2020].

the natural environment in which case you might want more of the traffic rumble. So the high pass filter would be off.

Atmos and sound effects are both great ways to build a world that your listener can become immersed in. Another strategy is to take advantage of the acoustics of different environments.

Using the Acoustics of Your Location to Set a Scene

Sound is always bouncing off surfaces and providing subtle information to the listener about the location of a recording. The creative minds at the BBC take full advantage of this when staging radio plays, as I observed when I visited their studio in Manchester in the UK. This studio contains walls with different surfaces that can be moved to form different arrangements. Scenes are set up so that it sounds as though the voice actors are in particular rooms. The studio has a specific prop for recording conversations inside a car and a special room full of acoustic treatment that sounds as though actors are outside. You might not have a world-class recording studio at your disposal, but you can get creative with the spaces that are available to you. If you are recording a scene for your fiction podcast that is set in a field, why not record it in an actual field? Earlier I advised against recording in a closet, but this could be perfect for a story about someone who has been taken hostage.

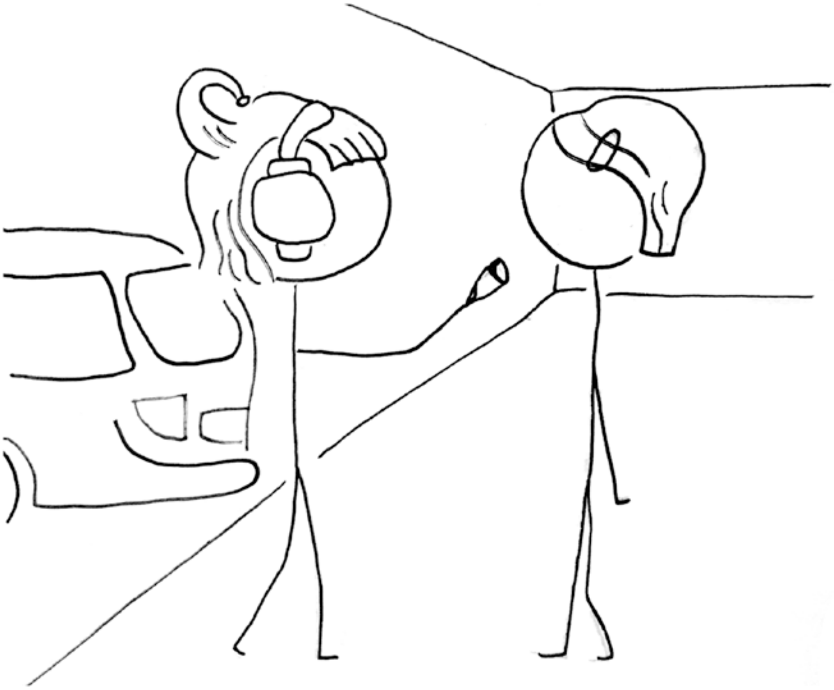

Figure 6-14. *An underground carpark is a distinctive acoustic space you could record in to set the scene for your next fiction podcast*

The acoustics of your recording environment is not just something to consider when making radio plays. When conducting an interview in the field, you should consider the way that acoustics helps with storytelling. As an example, grass absorbs slightly more sound than pavement. Next time you walk from a noisy city street into a park, listen to the reduction in background noise. Then concentrate on your body. How does the acoustic space of the park make you feel? Do you want this feeling in your podcast? Perhaps, but if you were recording a riot, it might be more exciting to record under an overpass so that the sound is thrown around in a chaotic manner and your interviewee has to shout into the microphone to be heard. These two examples are extremes, but they illustrate the point that you can use the acoustics of your recording to set a scene.

Recording Outside, but Actually Inside

Often when you're recording in the field, you're actually recording inside a building, which is pretty similar to recording at home. Even if you don't ever intend to record at home, the information in Chapter 5, "Recording Inside," will hold you in good stead for recording in any building.

Just like when you're at home, you can use your basic acoustic principles to locate what you think will be a good area to record in and then listen through your equipment. When you've found your ideal location, bring your interviewee to that spot.

When recording in a building, you might be able to bring a more complicated setup than you would typically bring to a field recording. If you can bring enough equipment to record each person on a separate microphone, this will greatly improve sound quality and make the editing process easier. It will be even better if you can provide separate headphones for each person.

If you're recording an event such as a conference, most of that equipment will already be there, and you will simply need to take a feed from the sound reinforcement system.

Plugging in to a Sound Reinforcement System

Many events such as conferences and concerts have a sound reinforcement system involving microphones and speakers set up so that a large audience can hear everything. A smaller event might have a *PA* (public address system) made up of a single speaker and a microphone. Whatever the size of the system, it's usually fairly straightforward to plug into an output and make a recording of proceedings.

I made a recording in this manner of social worker and veteran peace activist Margaret Pestorius speaking at an event connected to the Pine Gap trial. She was one of the people who was on trial, and she organized a lot of the legal defense. The recording I took has a definite flavor of PA to it, which sets a scene.

Audio 6-13 Recording through a PA

The first thing you will need to record at an event like this is good manners. Ahead of time, you will need to ask the organizer if you can record and broadcast their event. On the day, you will need to ask the sound engineer in charge if they mind your plugging into their setup. You will also need to ask the people speaking or performing how they feel about your recording and broadcasting their work on your podcast. At this stage, I would also recommend writing down the correct spelling of their name and what organization they're with so you can put it in your episode description.

Technically speaking, it's not difficult to plug your portable recorder into a sound reinforcement system. You will need to find a quarter inch (6.35 mm) output socket (which some people might refer to as a *jack*) as depicted in Figure 6-15. Alternatively, you might need an RCA output socket, which I will discuss shortly. The output will be labeled as something like "Aux" or "Out" or "Output," and the sound engineer is usually happy to help with locating it. On a big setup, look for the output on the mixing desk. If it's a small setup such as a microphone plugged into a single speaker, then look for the output on the back of the speaker.

Figure 6-15. *A quarter inch output socket (Simon Ashton/Shutterstock.com)*

You will need a cable with TRS connectors (as depicted in Figure 6-16) or TS connectors (Figure 6-17) on each end.

Figure 6-16. *A TRS connector (Anthony Maragou/Shutterstock.com)*

Figure 6-17. *A TS connector (Kr_photo/Shutterstock.com)*

Plug one end of the cable into the output and the other into an input on your portable recorder. The signal will come through one of your channels.

The Zoom H5 has combined XLR and TRS inputs, so you can plug the TRS connector into the same inputs that you would use to plug in the microphone (Figure 6-18). The TRS connector goes into the hole in the middle. The signal will come through either Channel 1 or 2 depending on which input socket you choose. Engage the button on the face of the recorder marked "1" or "2" to monitor your signal.

Figure 6-18. *If using a Zoom H5, plug the TRS connector into these inputs (photo courtesy of Zoom)*

A mixing desk gives out a *line-level* signal which is a lot stronger than a microphone-level signal. On many portable recorders, quarter inch input sockets are calibrated for a line-level signal, but not on the Zoom H5. If you're using the Zoom H5, you will need to turn the signal down using the pad function on your portable recorder.

On the Zoom H5, go to MENU ➤ IN/OUT ➤ In1/2PAD(-20dB) and select your input.

A line-level signal also has a different *impedance* to a microphone signal. It's not super important to understand impedance, except to know that if you plug a line-level signal into an input designed for a microphone-level signal, you might damage your equipment. If you're using a portable recorder besides the Zoom H5, check the manual to find out how you would plug a line-level signal into that unit.

If there's no quarter inch output socket on the mixing desk or PA speaker, you will need to find an RCA output, as depicted in Figure 6-19.

Figure 6-19. *An RCA output*

This output will require a different cable again. The cable will have a TRS connector on one end and an RCA connector, as depicted in Figure 6-20, on the other.

Figure 6-20. *An RCA connector (Yellow Cat/Shutterstock.com)*

Getting a direct recording through the PA or mixing desk is most effective for times when you're recording people talking. Often at small- to mid-level music events, not everything is being amplified. For example, the sound engineer might decide that the drums are loud enough on their own, and they don't need any microphones pointed at them. In this case, it might be better to record from in front of the stage using your portable recorder's inbuilt microphones. You will have to choose a location where you can hear well, but there's not too much noise from the crowd. If the mixing desk is located in a position to hear what's coming from the stage, then you can take a simultaneous recording of both the overall sound of the concert recorded with your inbuilt microphones and the feed from the mixing desk. At a conference, you can use this trick to simultaneously record both the presenters and the sound of the audience.

Plugging in to the sound reinforcement system will enable you to get a clear recording of an event, and it is especially useful for recording people talking.

Climatic Conditions

When recording outside, you're often battling the elements. Portable recorders and dynamic microphones are pretty sturdy, but you should exercise some care to keep them working well. Audio equipment generally doesn't like being wet or being too hot or being dropped. Don't leave it on the ground, even inside.

The Wind

One issue that can be super problematic when recording outside is the wind. Wind won't damage your equipment, but it can completely overwhelm your recording.

Audio 6-14 Recording on a windy day using the inbuilt microphones of a portable recorder

If you've ever tried to record outside using the inbuilt directional microphones of a portable recorder, you will already know that a slight breeze can frustrate your efforts. You can greatly improve the performance of the inbuilt microphones on a portable recorder by using a windjammer mic sock. I find that the RØDE Deadkitten (see Figure 6-21) works well.

Figure 6-21. *The Rode Deadkitten wind sock (photo by Karoline Morwitzer)*

A windjammer improves the performance of your inbuilt microphones. However, it's not completely effective against actual wind. This recording demonstrates the typical effect. You can gauge the strength of the wind by the sound of the screen door banging at the end of the recording.

Audio 6-15 Recording on a windy day using the inbuilt microphones of a portable recorder covered by a windjammer

Using your external microphone will improve the recording still more, especially if that microphone is a dynamic.

Audio 6-16 Recording on the same windy day using a portable recorder and external microphone

Using the external microphone with a windjammer will be even more effective.

Audio 6-17 Recording on the same windy day using a portable recorder and external microphone covered by a windjammer

In all of these recordings, I had the high pass filter turned on, partly because it's good to have it on for all recordings of human speech and partly because it will slightly improve recordings made in the wind.

There are other ways to deal with the wind. You can physically block the wind with your body or an object like a hat. You can step behind a building or another large object to move out of the wind.

Before I discovered the windjammer, I relied on the dual methods of using a microphone and blocking the wind with various objects. It's fairly acceptable as long as you have the patience to find a good spot. You can hear it in this recording I made at the Pine Gap trial with former Senator and committed peace activist Scott Ludlam. The recording is not bad for a noisy park on a windy day.

Audio 6-18 Using an external microphone and physically blocking the wind

Nowadays, I wouldn't record outside without a windjammer mic sock, but I still take sensible precautions such as moving out of the wind.

Dealing with the weather when recording outside can be challenging. Protect your gear and use the right equipment to make the best recording.

Note Taking

When you're recording audio in the field, it's useful to take notes to keep your audio organized for the editing process. You can take notes by speaking into your recording device or by using a pen and paper.

Before you start an interview with a person you have just met, it's a good habit to get some essential information. I recommend asking for the following:

- The interviewee's name and perhaps what organization they're with
- The spelling of the interviewee's name for when you want to write an episode description
- An affirmation that the interviewee gives permission to broadcast the interview wherever it is that you will be broadcasting

You might make note of the time and place of a recording or why the audio is important. If you're recording atmos or a sound effect, speak into your portable recorder at the end of the recording to take note of anything interesting.

If you're recording at an event, it might be inappropriate to talk, in which case you can take notes in your notepad. Write down the name of the file and what it is you're recording. If something exciting happens, write down a time reference and a short description.

Making notes of what you recorded and why will save you a lot of time during the editing process.

Summary

Recording outside is challenging, but your effort should be rewarded with interesting content, diverse perspectives, and stories told in vivid detail. The technical choices that you make will help you to produce high-quality recordings and to create a world that your listener will become immersed in. To deal with the unpredictable nature of recording outside, it pays to be prepared and to keep organized. Whether you're on the road searching for your next story, at your local street party, or finding the perfect sound effect for your sci-fi adventure, recording outside will take your work to the next level.

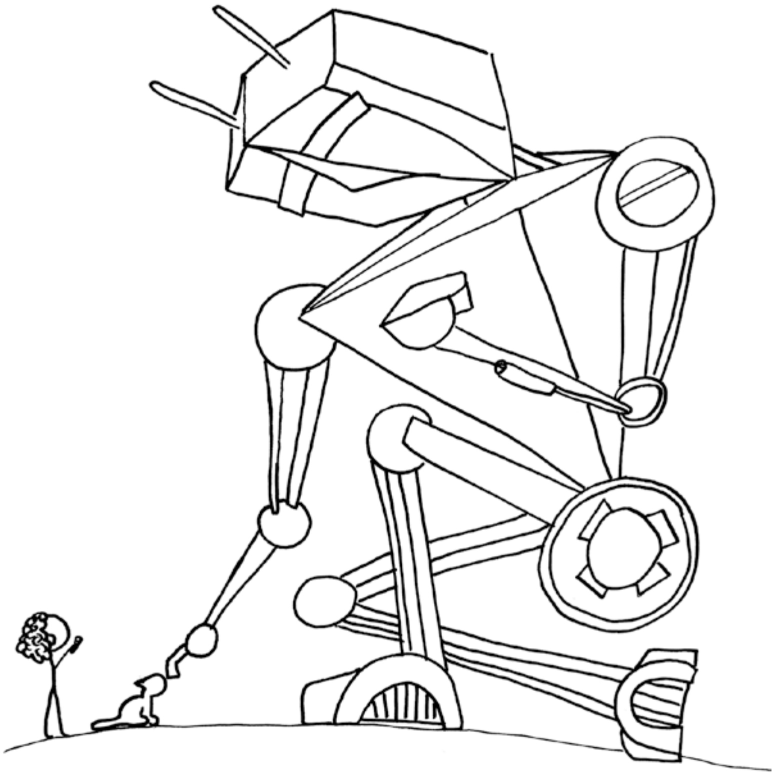

Figure 6-22. *Recording outside is worth the effort because a lot of interesting things happen outside*

CHAPTER 7

Recording Remotely

In 2020, the world faced the beginning of the COVID-19 pandemic. Everyone experienced this situation in different ways. Here in Australia, the government responded with an abundance of caution, particularly in Melbourne. As such, I got to live through one of the world's longest lockdowns. I don't think I'll ever forget the eerie silence on the first day of lockdown as an entire city ground to a halt. The events of 2020 threw up a lot of challenges and forced people to adapt to new ways of doing things. For those of us in lockdown, online video chat was a lifeline, but we still missed our friends and family.

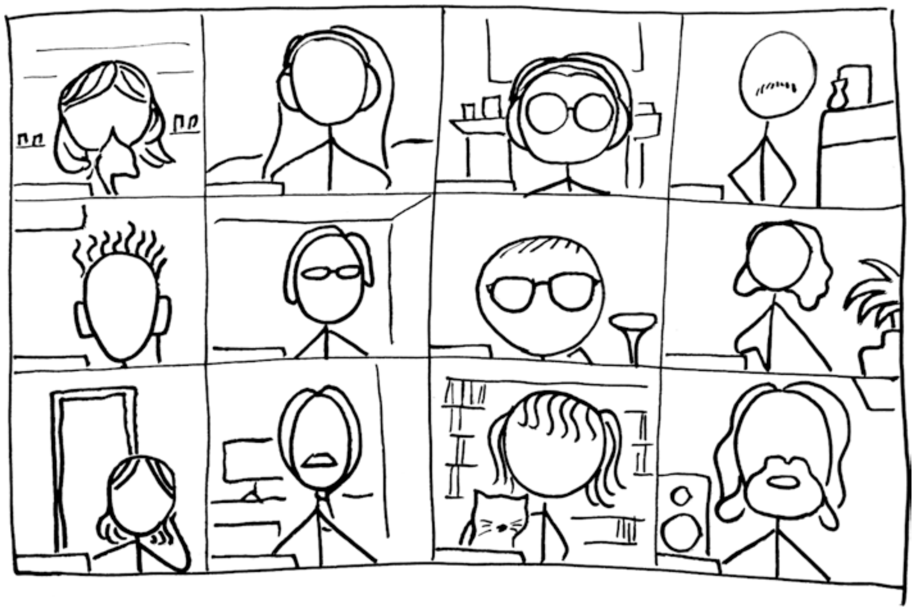

Figure 7-1. *Video chat has been a lifeline for those in lockdown*

C. M. Green, *The Podcaster's Audio Handbook*,
https://doi.org/10.1007/978-1-4842-7361-6_7

Remote recording predates the COVID-19 pandemic, but with radio stations closed or severely limited, it became an essential way for broadcasters and podcasters to reach their guests. As a result, the technology has advanced quickly in recent times, but it is still not quite as good as recording in the same room.

Recording remotely is convenient and enables you to feature a wider range of people on your podcast. However, when recording remotely, there are two areas in which the recording is compromised: the audio quality and the rapport that you can build with the other people on the call. I am going to take you through a recording platform, Zencastr, which addresses both of these issues. Zencastr relies on the guest having a computer and an Internet connection, so I will also discuss how to record a phone call.

This chapter contains audio files I have made to demonstrate the quality of different recording methods. I have cut and moved audio, added fades, and adjusted the levels. The programs I am demonstrating have added their own processing.

Before I start, I'd like to look at some issues you will face in remote recordings.

Issues Faced When Recording Remotely

When recording remotely, you don't have a lot of control over your guest's audio setup. I'm sure we're all familiar with the sound of the beginning of a Zoom meeting. "Sorry? What? Your microphone's not turned on… I think you're going to have to move a bit closer…." If you are creating a remote recording, then the best you can do is discuss equipment ahead of time and also leave time to troubleshoot at the beginning of every recording session.

In addition to a lack of control over the guest's recording equipment, audio quality is compromised when audio is sent over the Internet or a telephone line. You can hear that it sounds hollow and muffled.

That's because a lot of data needs to be removed to send audio over the Internet or the phone in a timely and efficient manner. The same sorts of algorithms that are used to compress audio into mp3s are used to compress audio to stream it over the Internet. Even if you can hear the speech pretty well from a typical VoIP (Voice over Internet Protocol) call such as a Zoom chat, your audio will go through additional compression when you turn your completed show into an mp3 and then more again when your podcast is streamed through a distribution service such as Spotify. After all that, someone may well listen to your podcast through low-quality earbuds on the train. By the end of the process, you will have only a fraction of the audio data that you began with, so it is best to begin with the largest amount of data possible. To avoid VoIP compression when recording a podcast, you can record the audio at both ends of a call and combine the files later. This is called *double-ended recording*. Zencastr is an online audio recording platform that makes this easy.

Another issue that plagues VoIP recordings is Internet dropouts, but these are not an issue when recording with Zencastr. As well as making double-ended recordings, Zencastr sends audio through VoIP so that everyone in the session can hear each other. If you're conducting an interview using Zencastr and you hear the audio dropping out, you need not be unduly concerned about the recording. The VoIP stream might be faltering, but everyone's respective computers are still recording their own audio files at their end. In this way, Internet dropouts are circumvented.

Audio quality is not the only thing that is compromised in a remote recording. Physical distance robs a conversation of context and of social cues such as body language. This makes it more difficult to build a rapport with your guests. While it's not perfect, a video call will restore some of what is missing and help you get to know people on a level you might not experience with a phone call. The global pandemic has been a hard time, but an enjoyable element of all the video calls has been meeting so many people's pets. Zencastr provides a video platform that enables you to see your guest while you're recording high-quality audio.

Recording remotely is not all bad. The major benefit is that you can access guests who live a long way away from you. It is more convenient than traveling to your guest, but if you find yourself wanting to record over the Internet when with a reasonable amount of effort you could record in person, think of the compromises to quality.

Figure 7-2. *Video chat enables users to see more of each other's social contexts and their pets*

Recording Remotely with Zencastr

Recording in person is better than recording remotely, but if that option is unavailable to you, then the remote recording platform Zencastr is the next best thing. Zencastr is packed with features, but I will discuss a few essentials and leave you to explore the rest.

Zencastr is a platform that enables you to create a high-quality remote recording through a web browser on a computer. You simply create a session in your web browser, invite your guest, and make a recording. You don't even need to download a program. At the end of the session, your recording is saved and sent to a *cloud storage service*, such as Dropbox or Google Drive. Ordinarily, you'd store files on your own computer, but with cloud storage, your files are sitting on another computer called a *server* in a warehouse somewhere else in the world, accessible to you over the Internet. If you're happy to receive your audio files in the mp3 format, then the platform is free, but I recommend paying for Zencastr Pro so that you can have the recording saved in the wav format. I'll go into how to set up Zencastr in more detail shortly. For now, I want to demonstrate the quality of the recordings.

Here's a recording I made using Zencastr with my friend Tuffy. Tuffy's a documentary filmmaker so they have a pretty decent audio setup. For the sake of comparison, Tuffy and I were simultaneously recording over Zencastr and Zoom. Listen for the difference in quality between the audio that's been sent over VoIP (Audio 7-1) and audio that's been recorded with Zencastr (Audio 7-2). You can hear the greatest difference in Tuffy's side of the conversation because, in the Zoom example, that's the side that's being sent over VoIP. The Zencastr recording sounds more clear and open. The Zoom recording drops out, whereas the Zencastr recording is consistent throughout.

Audio 7-1 Recording with Zoom

Audio 7-2 Recording with Zencastr

While Zencastr is geared toward high-quality recording, the platform also caters to situations where audio quality might be compromised.

High Quality Meets Low Quality

The creators of Zencastr appear to be obsessed with high-quality audio, which is something I can deeply relate to. They recommend that you use headphones and a microphone to get the best out of their platform, but they recognize that you might not always meet this ideal.

Headphones reduce the amount of processing that is added to audio that is being recorded remotely. When you're making a call using computer speakers, the microphone you're using picks up everything that the speakers are putting out. This is why you sometimes hear an echo in a call. Most of the time, the echo is filtered out by something called *acoustic echo cancellation*. Acoustic echo cancellation is effective, but it can compromise audio quality. The creators of Zencastr recommend headphones to avoid this compromise.

Similarly, the creators of Zencastr recommend that you use an external microphone with their platform. I agree with this approach because the microphones that are built into laptops are not well suited to the task of creating a podcast. Headphones and a microphone will enable you to create the highest-quality recording with Zencastr.

Sometimes it is not possible to use an audio setup that is conducive to high-quality recording. If your guest doesn't have an external microphone or headphones, it's not the end of the world. Zencastr adds some processing to the audio for those times when compromises have to be made.

I conducted a test (Audio 7-3) with my friend Ian in poor recording conditions.[1] On the day of this recording, Ian didn't have a microphone or even headphones and was sitting in a cafe. If you listen closely, you can hear that Zencastr turns down the background noise on Ian's track a little when he's not speaking. You can hear that the acoustic echo cancellation is doing its job. Despite the conditions, the audio quality is a lot better than you'd expect.

Audio 7-3 Recording remotely using the inbuilt speakers and microphone of a laptop in a noisy cafe

[1] Special shout-out to Ian for testing about a dozen methods of remote recording with me for this book.

Audio that has been compromised in this way particularly benefits from being recorded at both ends in the wav format, to make up for a lack of quality audio data.

Figure 7-3. *Using Zencastr without a microphone or headphones is like using the Batmobile to pick up groceries: still awesome*

Zencastr has been designed for recording high-quality audio files and contains features that will improve recordings on days when compromises must be made. Now that you understand some of the key features, let's look at how to operate Zencastr in more detail.

Setting Up Zencastr

Zencastr is easy to use once you understand how it operates.

Zencastr runs only on a computer and not on a phone or tablet. This is because it makes large audio files and so needs to have access to a lot of temporary storage.

The first step to using Zencastr is to sign up to an account at **www.zencastr.com**. There's no need to download a program; you simply create the recording through your web browser. Currently, Zencastr only supports Google Chrome, Brave, and Edge. I would recommend using a browser other than Google Chrome, because on some computers, Google Chrome will override your audio routing if it thinks you've made a mistake. This can lead to unpredictable outcomes.

When you log in to Zencastr, the first screen that you see is your dashboard. This is the screen where you can access any files you have already recorded.

Before you make your first session, you will need to connect your account to either Google Drive or Dropbox which are cloud storage services. Any files that you record will be available to download from the cloud storage service that you choose. You connect Zencastr to a cloud storage service through the dashboard as in Figure 7-4.

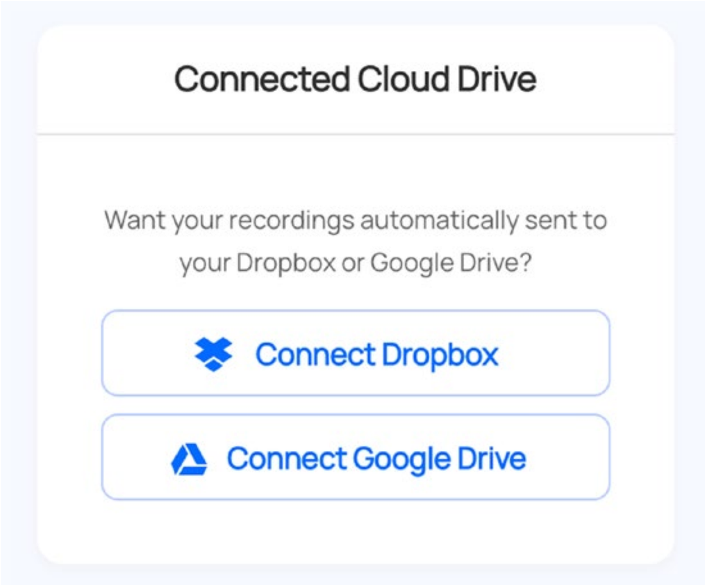

Figure 7-4. *Choose a cloud drive*

Setting up an account with Google Drive or Dropbox isn't hard. If you have a Gmail account, you already have access to Google Drive. You can find it by logging into your Gmail and looking for the following icon in the top right-hand corner.

Figure 7-5. *Click this icon to find Google Drive*

A menu will come up and you will find a number of services, including Google Drive. The icon for Google Drive is as follows.

Figure 7-6. *The Google Drive icon*

After connecting your account to cloud storage, you should check that you're recording in the wav format as in Figure 7-7. You will need to pay for a subscription to access this option, but I think that is money well spent.

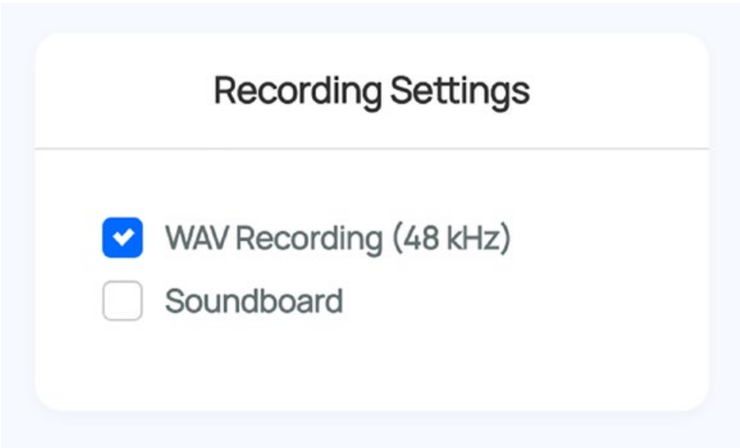

Figure 7-7. *Check this box to record in the wav format*

Making a Recording with Zencastr

You are now ready to make your first *episode*, which is a session within which you will meet your guests and create a recording. Creating an episode with Zencastr is straightforward because there are no time limits on the episodes. You create the episode, and then it stays up on the platform until you take it down. Your episode can contain one or more recordings. To create the episode, go to your dashboard and click the button that says "Create New Episode" which you can see in Figure 7-8.

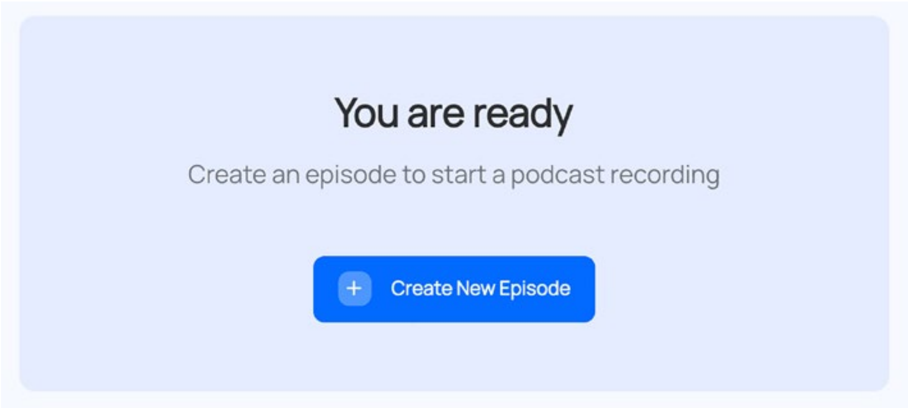

Figure 7-8. *Click this button to create a new episode*

A screen will come up offering a few options. Choose the one labeled "Record Audio Only. Show Video." This should take you on to your episode page.

From the episode page, you will need to send an invitation to your guests to join you. Figure 7-9 shows the button you select to invite a guest. This button is quite small and located at the top of the screen.

Figure 7-9. *Click this button to invite a guest*

Clicking the invitation button will open a window that allows you to invite guests via email. You can invite any number of guests to your session. The email invitation includes a link to the episode and a checklist on ways the guests can maximize their audio quality.

After you have invited your guest, your episode page should look something like Figure 7-10. This is me and my friend Ian in our episode.

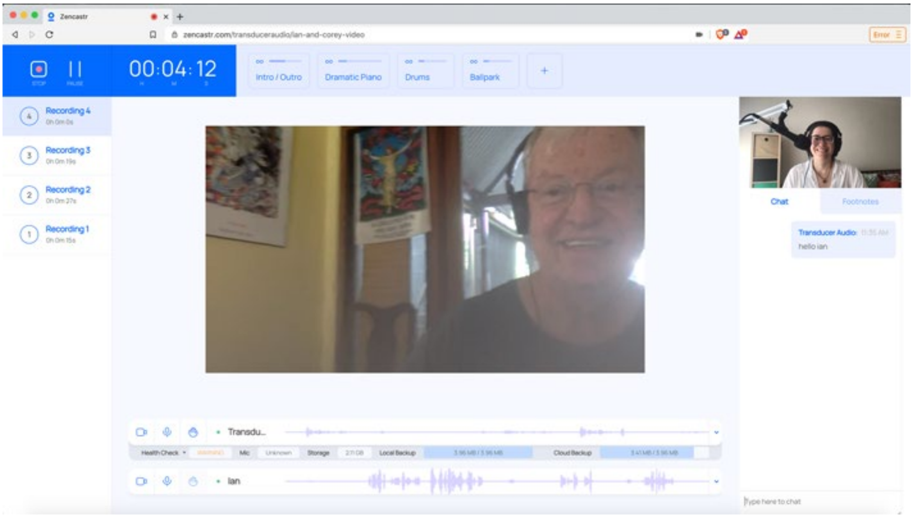

Figure 7-10. Ian and Corey in our episode page

So you've set up your episode and you're chatting with your guests. It's important to remember there are two things going on with Zencastr. Zencastr makes separate high-quality recordings of every person on the call. It also transmits low-quality audio and video using VoIP so that everyone present can hear and see everyone else. When you start your session, you're enabling VoIP but not the recording. That means that once your guests have arrived, you can chat with them, but you will need to manually start the recording as per Figure 7-11.

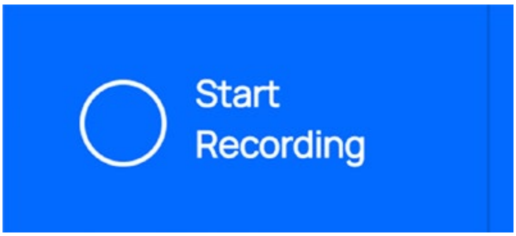

Figure 7-11. Click this button to start your recording

However, before you start your recording you should first perform a sound check.

Performing a Sound Check in Zencastr

In Chapter 4, "Getting a Good Take," we discussed performing a sound check. When you are making a remote recording, you will need to perform that same sound check, and there will be a few extra issues to consider.

Due to the nature of remote recording, everyone in the session will need to sort out their own audio, but with your guidance. You will need to listen to your guests and instruct them on issues such as microphone technique and whether they have set their equipment to a good level.

Once you have done this, you can address the issues unique to remote recording.

Resolving Echo

An issue you might encounter in a remote recording is the sound of an echo as you speak. I briefly touched on this problem earlier, but I'll go into more detail now. In a remote recording, echo is caused when Person 1 speaks into their microphone, the audio travels through the Internet to the speakers on the computer of Person 2, and then the sound is picked up by the microphone of Person 2 and sent back to the headphones of Person 1.

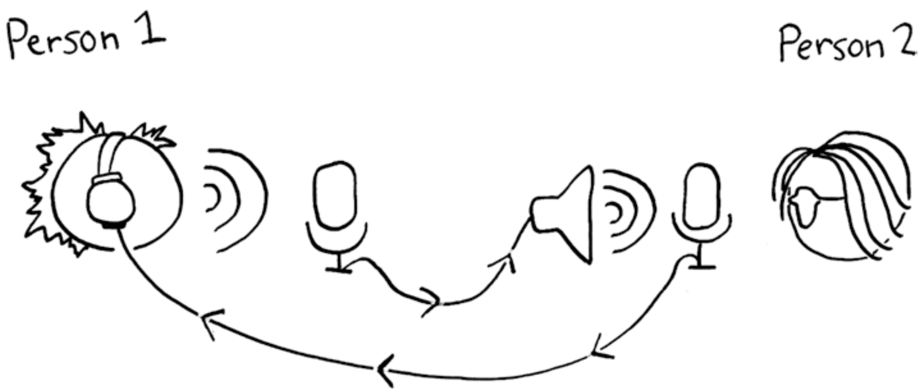

***Figure 7-12.** Echo occurs when the sound Person 1 makes is sent back to them via Person 2's audio equipment*

This means that even though the problem is happening due to the equipment of Person 2, it's affecting Person 1. You need to listen out for echo and ask your guests whether they are experiencing this issue due to your equipment.

To prevent echo, Zencastr recommends that everyone uses headphones to isolate the output from the input. So in Figure 7-12, the output is the sound coming from the computer speaker of Person 2, and the input is Person 2's microphone. The output is feeding into the input. If Person 2 was wearing headphones instead of using a computer speaker, then the output would not be feeding into the input, and Person 1 would not hear an echo.

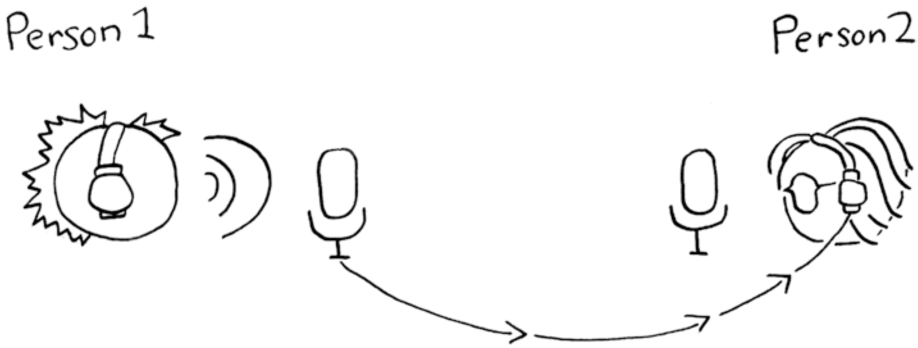

Figure 7-13. *If Person 2 wears headphones, then the sound that Person 1 makes will not be sent back to them*

The simple way to resolve echo is by going into the settings menu of your Zencastr episode and enabling acoustic echo cancellation as in Figure 7-14. However, acoustic echo cancellation can compromise audio quality, so it's better to try to troubleshoot it first.

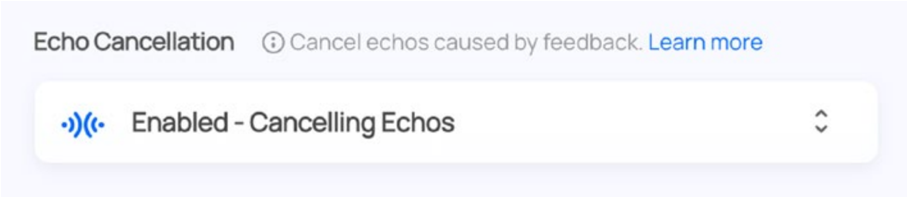

Figure 7-14. *Enable echo cancellation in Zencastr*

Sometimes you are in a situation where everyone is using headphones and there's still echo. Theoretically, when you plug your headphones into your computer, it should disable the computer speakers, but it doesn't always work. So, if you are hearing an echo, then you need to ask the other session participants to check whether their computer is sending audio to their computer speakers as well as their headphones. If this is the case, the person who has audio coming out of their computer speakers needs to go into the settings of their web browser and manually route the output of the computer audio to their headphones. I have found that on many computers Google Chrome will override this routing and still send audio out of the speakers. In this case, you can either change which browser you're using with Zencastr, or the host of the Zencastr episode can enable acoustic echo cancellation.

Headphones are usually effective at isolating audio output from the microphone, but some headsets fail at this. This is because the microphone on the headset is close enough to the headphones to pick up audio. This is a particular problem with the little Bluetooth headsets. You might improve this by turning down the volume of the headphones part of the headset. If all else fails, acoustic echo cancellation is always waiting for you.

Resolving Lag

Another issue that can be distracting during a remote recording is the presence of a high amount of latency, which is often referred to as *lag*. This can cause the video and audio to lose synchronization. If there's a long lag, it can interrupt the flow of the conversation and cause people to accidentally talk over each other. Lag can significantly compromise the quality of an interview.

There are a number of things that might be slowing down your computer or Internet connection and so causing lag. Before you start recording, go to the settings menu in the Zencastr episode and uncheck the box marked "Record in maximum quality" as in Figure 7-15. This will reduce the quality of your video, but video is not important for podcasting.

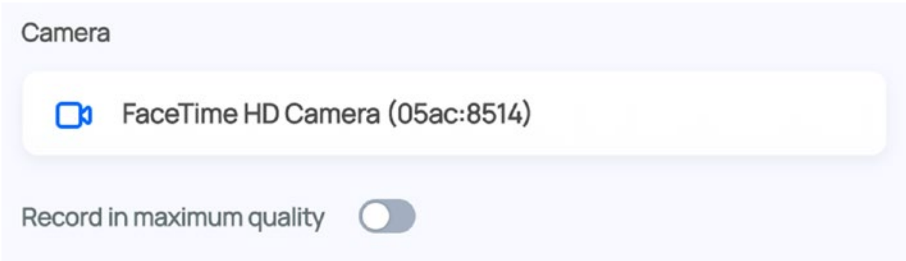

Figure 7-15. *Uncheck the box marked "Record in maximum quality"*

Zencastr recommends that everyone in a session shuts down unnecessary programs on their computer to reduce lag.

If you've tried your best and there's still lag, then simply turn the video off for the whole session.

If you are still experiencing lag in your audio when you have the video off, then you and your guests will have to start troubleshooting the quality of your Internet connection.

Setting Levels in Zencastr

You need to pay particular attention when setting levels in Zencastr. Even though Zencastr creates high-quality files in the wav format, the audio in these files has been encoded at 16-bit, as opposed to the 24-bit encoding that is preferable for podcasting. This is due to the need to create smaller files that can be sent through the Internet more quickly. To compensate for the smaller file size, you need to pay particular attention to setting your microphone input level, because there is less room for error.

Unfortunately, Zencastr gives you very little information with which to set your microphone level. To see any kind of useful information about your level, you will need to select "Start Recording" as in Figure 7-16.

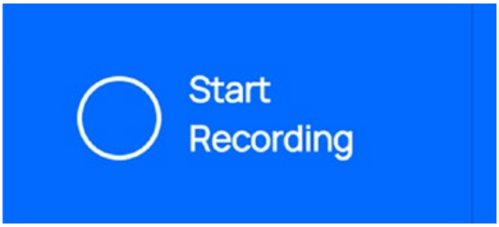

Figure 7-16. *Click this button to start your recording*

Once you have started your recording, a purple line will appear below the video. If you speak into your microphone, a representation of a sound wave will appear.

You will need to adjust your gain until you reach a healthy level – that is, a level in which you are recording an adequate amount of audio data but not peaking.

Figure 7-17 is what a healthy level looks like in Zencastr.

Figure 7-17. *How a healthy level looks in Zencastr*

You should avoid setting your level so high that it is peaking. Figure 7-18 is what audio looks like in Zencastr if it is peaking.

Figure 7-18. *Audio that is peaking in Zencastr*

If your level looks like Figure 7-19, you are not recording enough audio data and need to turn the gain up.

Figure 7-19. *Recording at a level that is too low in Zencastr*

Some audio equipment, such as headsets, doesn't come with a gain control. In this case, you will need to adjust the gain through the sound settings of your operating system.

To do this on a Mac, go to Apple ➤ System Preferences ➤ Sound ➤ Input. Then adjust the "Input volume" fader as in Figure 7-20.

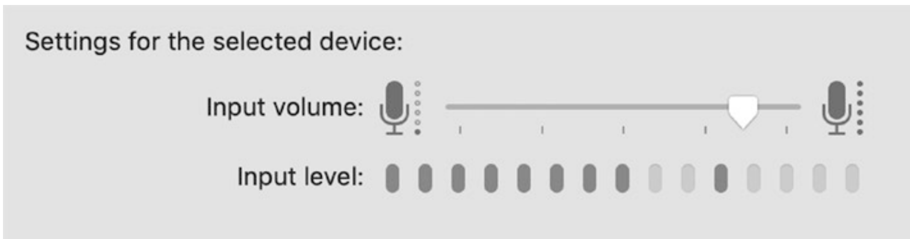

Figure 7-20. *Adjusting the input volume on a piece of audio equipment through the system settings on a Mac*

On a PC, go to Start ➤ Settings ➤ System ➤ Sound. Scroll down to "Input." Click "Device Properties." Click "Start Test." Talk into your audio equipment, and then adjust the volume fader. A line should appear underneath the fader representing the input volume. Figure 7-21 is a screenshot of the test in action.

Figure 7-21. *Adjusting the input volume of a piece of audio equipment through the system settings on a PC*

You will have to check the level for all of your guests, but they will have to adjust their own equipment according to your instructions.

After you have sorted out your levels and other audio issues, you can create a remote recording.

Uploading the Recording

It's advisable to stop recording in Zencastr as soon as you have finished your interview. The wav files created by Zencastr are huge, and they take some time to be uploaded to cloud storage. If you want to chat with your guest after the recording has finished, then you can do that while the file is uploading. You need to remain logged in to Zencastr on the episode page until the file that is stored on your computer has uploaded to cloud storage, and each of your guests must do the same. It would be wise to advise your guests of this just before you stop the recording. Zencastr sends each user a message, as in Figure 7-22, when their file has been uploaded successfully.

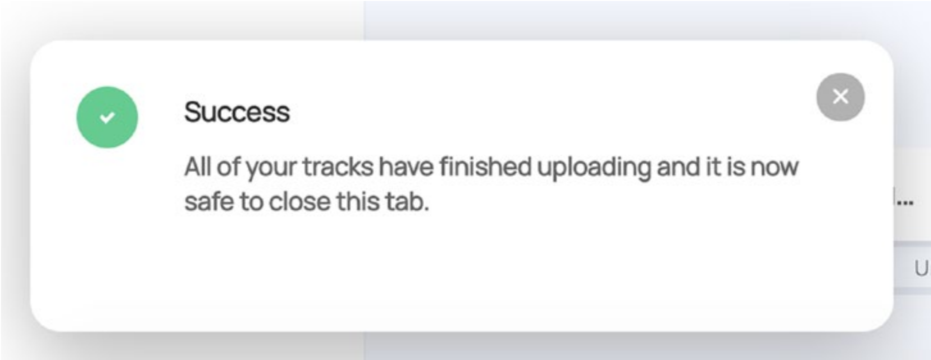

Figure 7-22. *When you see this message, you can close your browser window*

If a person accidentally closes their browser window before their file has uploaded to cloud storage, then it is worth them logging back in to the episode page to see if the temporary file is still on their computer. If you're lucky, the upload will start again.

After all of the files have uploaded, you will find your recordings in a folder marked "Zencastr" in the cloud storage service that you chose, ready for you to download.

Zencastr is the best way to make a remote recording, but it is only available to people who have an Internet connection and a laptop. If your guest only has a telephone, then you will need to explore other options.

Recording a Phone Call

You might want to interview a person who has a telephone, but doesn't have a laptop. Recording a phone call is more complicated than recording a call over the Internet; the audio quality is inferior, and there's no option of video chat. This method of remote recording is undesirable, but still useful for reaching certain people.

Figure 7-23. *You may want to interview a person who is undertaking a solo unicycle journey to raise awareness of clowns and therefore does not have a laptop*

You can record a telephone call by *routing* audio from Skype to your audio editing software. Routing is a term that describes moving audio from one place to another. Skype is a well-known video chat program. It's free to download this program and to create an account. I have chosen this program because you can use it to call a telephone from your computer for a small amount of money. The Skype telephone call service combines VoIP technology with the telephone network. This results in compromises to audio quality, and lag. Lots of lag.

Audio 7-4 is an example of a Skype telephone call that I recorded with my friend Riah. At the beginning of the call, you can hear the Skype calling tone, which then switches to a telephone calling tone. This is the two networks connecting. Riah is at the recording end of the call, and he sounds present and clear. I am on the telephone and so sound quite distant and muffled. You'll also notice that there's quite a lot of lag. Lag can easily be edited out, but it interrupts the flow of conversation.

Audio 7-4 Calling a smartphone from Skype

There are a few ways to route audio from Skype to Audacity. There's virtual routing programs such as VoiceMeeter, Loopback, and Soundflower. Physical routing is more straightforward, but there's still quite a bit to consider.

Routing a Phone Call from Skype to Audacity

Routing can be hard to get your head around. I find that it always helps me to draw a diagram. The routing setup that I am about to go through looks like Figure 7-24.

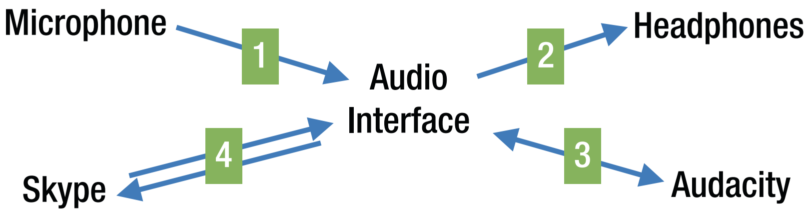

Figure 7-24. *Routing diagram for recording a Skype call through Audacity*

To explain my diagram

1. The microphone goes into the audio interface.
2. The audio interface outputs sound to headphones.
3. Audio is sent to Audacity from the interface via USB cable for recording. It's also sent from Audacity to the interface by the same USB cable for playback.
4. Skype comes out of an output on the audio interface and is looped back into an input with a cable.

This might look complicated, but it's not really that hard to set up. I'll take you through the steps in a methodical way, but first we need to discuss feedback loops.

When routing audio using any method, you need to be mindful of creating feedback loops. A feedback loop occurs when an output feeds to an input that then feeds to that same output, so they're linked circularly. In between the input and the output is a source of amplification, so any sound that enters the system loops around between the input and the output becoming progressively louder until you hear an unearthly screech.

Figure 7-25 is a feedback loop at its most basic. It illustrates a scenario where the host of your local bingo night is calling numbers through a small PA system with one speaker and one microphone. The speaker and the microphone are linked once with cables via a source of amplification, and they're linked a second time when the host accidentally points the microphone at the speaker while proclaiming "legs eleven." Your routing may be more complicated than the setup at bingo night, but a feedback loop is always essentially caused by this same issue.

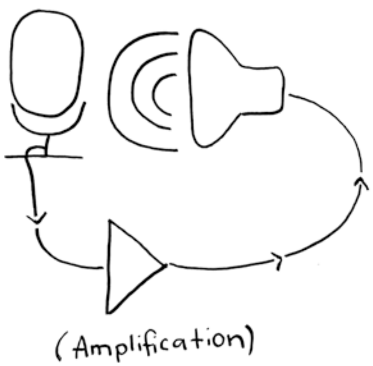

Figure 7-25. *A basic feedback loop. The speaker and the microphone are linked using cables via a source of amplification. A loop is created when they are linked a second time by the microphone picking up the speaker output*

When routing, take off your headphones before making any changes in case you inadvertently cause a feedback loop. If you're not careful, the unearthly screech may be blasted straight into your ears which is unpleasant and could cause hearing damage.

Now that we have that out of the way, we can begin the process of routing Skype to Audacity.

Step 1: Sound Check Your Microphone

To start off, plug your microphone into input 1 of your audio interface as usual. Also, plug your headphones into the headphones socket of the audio interface.

Direct monitoring is a feature that enables you to monitor what audio is coming through the audio interface input channels directly - that is without the audio going through your computer first. When you're recording a Skype call, you don't want to direct monitor your microphone signal, because this will set up a feedback loop. But you do want to direct monitor while you're setting up your microphone so that you can do a sound check.

On most audio interfaces, direct monitor is labeled as such. You will always hear what's coming out of the computer, and the "direct monitor" switch will toggle whether you can hear the channel inputs as well. The *Presonus Studio 24c* gives you the ability to fade between the input channels and the computer playback. To turn on direct monitoring, turn the knob labeled "Mixer" (marked in Figure 7-26) all the way to "Inputs."

Figure 7-26. *The knob to control direct monitoring on the Presonus Studio 24c*

Now go into Audacity. You will need to assign the input and output of Audacity to your audio interface.

In Audacity, you don't have to go to the Preferences menu to set the input and output; you will find drop-down menus on the main screen. The input is marked with a microphone, and the output is marked with a speaker. Set the input and the output to your audio interface as in Figures 7-27 and 7-28.

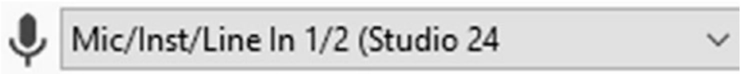

Figure 7-27. *Set the input in Audacity to your audio interface*

Figure 7-28. *Set the output in Audacity to your audio interface*

When you're recording or monitoring your microphone through Audacity, you want to prevent Audacity from sending audio back to the interface. This is partly to prevent hearing an unnerving lag between when you speak and when you hear yourself. It's also going to prevent a feedback loop later in this process.

So on a PC, go to Edit ➤ Preferences ➤ Recording, and untick the box marked “Software playthrough of input.” On a Mac, go to Audacity ➤ Preferences ➤ Recording, and untick that same box.

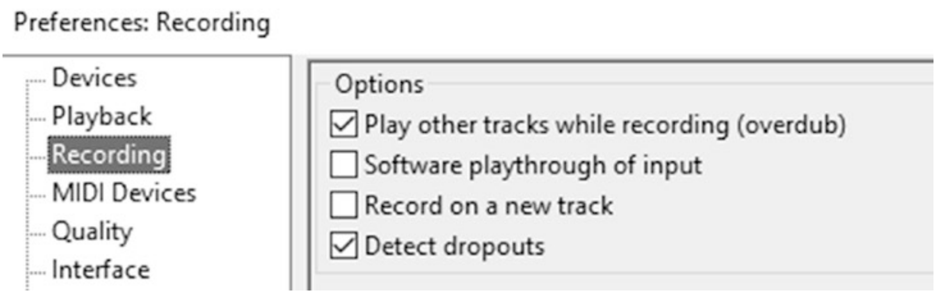

Figure 7-29. *In Audacity’s Preferences menu, untick the box marked “Software playthrough of input”*

Now create two mono tracks. Label the first “Microphone” and the second “Skype.” Perform a sound check for your microphone signal as set out in Chapter 3, “Getting a Good Take.”

When you have finished your sound check, turn off the direct monitoring, so that your routing doesn’t result in a feedback loop. On the Presonus Studio 24c, turn the knob labeled “Mixer” all the way to the side marked “Playback.” You won’t be able to monitor your own microphone signal during the Skype call, so it’s important that you spend time getting the sound check right.

Monitoring your own audio while recording helps maintain quality, so if this is important to you investigate audio routing software programs such as Loopback, VoiceMeeter, or Soundflower.

The method of remote recording we are discussing will capture system sounds, such as a ding your computer might make when you click the wrong button. If you don’t want system sounds in your recording, turn those off now.

Step 2: Routing Skype

If you haven't already, download Skype, install it, set up an account, and put some money on your account.

To route your interface to and from Skype, go to the Preferences menu and then onto "Audio & Video." Set the input (labeled as "Microphone") and output (labeled as "Speakers") to your interface as in Figure 7-30.

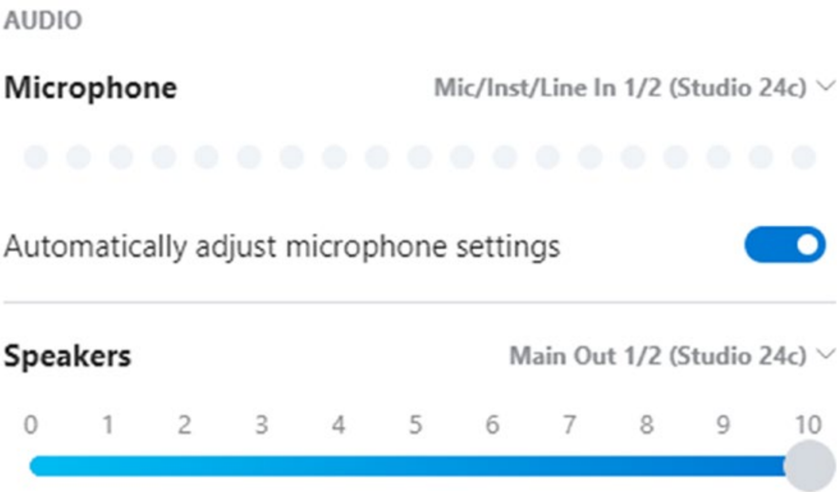

Figure 7-30. *In the Skype Preferences menu, set the input (Microphone) and output (Speakers) to your audio interface*

Your microphone signal will go into Skype, but not out of Skype because Skype only sends the other side of the conversation through the output.

Now the Skype signal is coming through your audio interface you want to take that signal and feed it back into input 2 on your interface so that you can record it on Audacity. For this, you will need a cable with TRS connectors (Figure 7-31) on each end or with TS connectors (Figure 7-32) on each end.

Figure 7-31. *A TRS connector*

Figure 7-32. *A TS connector*

Take the TS or TRS cable, and plug one end into one of the outputs of your audio interface (Figure 7-33) and the other end into input 2 (Figure 7-34).

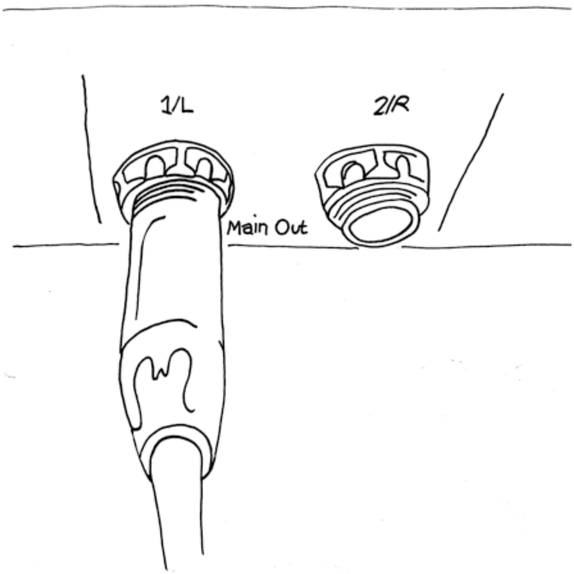

Figure 7-33. *Plug one end of a TS or TRS cable into the left output of your audio interface*

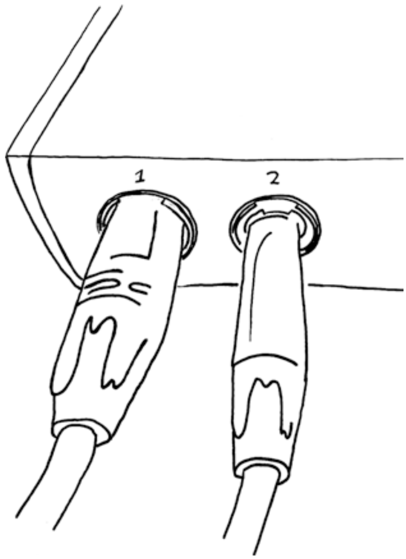

Figure 7-34. *Plug the other end of the TS or TRS cable into channel 2 of your audio interface. Channel 1 has the microphone plugged into it*

Set the main output level of your audio interface to 0dB (which is the maximum value). On the Presonus Studio 24c, the output level is marked as "Main." Turn the input gain of channel 2 to about halfway for now. You will need to adjust it later when you have your guest on the line.

Step 3: Making Your Skype Call

Now go back to the Skype window, and make a telephone call by navigating to File ➤ New Call. You will find the option to bring up a dial pad on the right-hand side of the search box as in Figure 7-35.

Figure 7-35. *The dial pad option is to the right of the search function*

Here, you can input a telephone number and make a call.

Inform your guest that you're about to start the recording. Return to Audacity, hold down the shift key, select both tracks, and press record.

Set the level of the Skype call using the gain knob on input 2 on your audio interface.

The audio quality of a phone call is nowhere near as good as the audio quality of Zencastr, and there's no video chat option. However, this method will allow you to interview people who have access to a phone, but not a computer with an Internet connection.

Summary

Recording remotely enables you to access a wider range of people, but can compromise both audio quality and the quality of the interview. The best option is to record in person, but if you are recording remotely, then choose Zencastr.

It is also possible to record a telephone call by routing the audio from Skype to your audio editing program. This will allow you to access people who have a phone but don't have a computer with an Internet connection. This is more difficult than using Zencastr, the audio quality is compromised, and there's no option for video chat, so consider it the last resort.

Now that you understand remote recording, you will be able to choose the best option for your circumstances and record a wide variety of people.

CHAPTER 8

Editing

I once worked with a woman who would constantly tell me stories about her cat. Generally, I find stories about people's pets quite tedious, but these stories had me in stitches. I'd come into the office anticipating the next instalment of the banal adventures of this unfortunate creature.

Of course there was nothing special about the cat; it was my co-worker who was the master storyteller. Timing, pacing, even sound effects – these stories were a work of genius.

Figure 8-1. *The unfortunate cat*

When you're making a podcast, you have more tools available than if you are talking to your co-worker over the photocopier, but nevertheless, storytelling is the heart of the process. The art of telling a story is knowing what to leave in and leave out, it's timing and pacing, and it's knowing how to arrange information for the maximum impact. In this chapter, I will be discussing techniques for arranging information using audio technology. And if you want to draw someone into your world, you'll need to make

C. M. Green, *The Podcaster's Audio Handbook*,
https://doi.org/10.1007/978-1-4842-7361-6_8

sure that there are no niggling audio issues that distract from the content, so I will be covering loudness and fades. While editing a podcast requires technical skills, you can ultimately think of the technical choices you make as creative choices in a storytelling process.

Sound engineers spend years learning and refining editing techniques, but in this chapter, I will only be covering the basics and I am not going to delve deeply into repairing audio. Instead this book has focused on getting the recording stage right, because that is easier and more effective than fixing bad audio. If you start with a good recording, you can make a pretty solid podcast with just these editing techniques. If you start with a bad recording, then a sound engineer can probably use their skills to turn it into a pretty OK podcast. If you start with a good recording and have a sound engineer edit it, then you can create a polished product. Think about the outcome that you're after and decide where to concentrate your resources. After you've got the basics down, you should be able to better engage with existing sound engineering resources to learn about audio repair.

Your Editing Setup

To edit a podcast, you will need audio editing software and some kind of hardware that will enable you to listen back to your audio. The software that you use to edit your podcast is dependent on the level of editing that you plan to undertake, and the hardware should be determined by how you expect your audience to be listening to your work.

There are many computer programs that you can use to edit a podcast. Audacity is a good choice for creating a basic podcast. As I stated earlier, it's free, it works on Windows and Mac, and it's adequate for basic audio editing. It's reasonably easy to learn how to use Audacity, but it does have some limitations. Most audio editing programs will let you apply audio effects and processing, such as equalization and compression, while your audio file is playing. This means that you can hear the changes you're making in real time. Audacity does not support this option, and so it is very difficult to apply effects and processing skillfully in this program. If you are planning to use effects and processing on your audio, then you will need to use another program.

Reaper is an affordable audio editing program that is a good option for a nonprofessional. It offers everything that you would expect from a modern audio editing program, including the ability to adjust processing and effects as you listen to your audio file.

At the professional end of the scale are programs like Pro Tools. Pro Tools is expensive, complicated to learn, and it does a great job. As a fancy pants sound engineer, this is my choice, and it is a great choice for a person who wants to master audio editing.

You will need to make some decisions about the hardware you use to listen back to your podcast during the editing process. You can choose headphones, your computer speakers, or high-quality speakers. Your choice should be determined by what equipment you expect people to use to listen to your work. For example, you might be editing using high-quality speakers, and you might include some spooky ambient music made up of very low-frequency sounds. If most of your audience is listening back to your work through a smartphone, they will not get the full effect of those low-frequency sounds.

It's good to primarily edit a podcast using quality over-ear headphones because these will give you a lot of detail. Headphones are also a good choice because many people listen to podcasts using headphones, especially if they are listening while using public transport. This could be a large section of your audience if you are making a morning news podcast. I discuss headphones in Chapter 2, "Gear Part 1" (section "Headphones").

During the editing process, you should sometimes listen to your work on your computer speakers as well, especially when you are layering different elements such as music and sound effects with speech. This is because the speakers that come with your computer are a good proxy for how your podcast might sound when someone is listening through the speaker of their smartphone. Audience members who are listening at home are likely to listen to your work through a smartphone. If your computer speakers are really bad and you can still hear all of the elements in your podcast, then your editing has been successful. If I am layering elements, I will listen quietly on my computer speakers. I feel really pleased with myself if a truck goes by and I can still hear everything clearly.

If you anticipate that people will be listening to your work through quality speakers, then you too should also spend some time in the editing process listening through quality speakers.

Now that you've made some decisions around editing software and hardware, you might be excited to get started, but to save yourself from tears, you should back up your recordings first.

Backing Up Your Audio

You have spent time and painstaking effort capturing your audio, and now it's time to start editing. Before you begin, you should make backups so you don't lose any valuable audio data in the editing process.

Figure 8-2. *Make sure you can call for backups in case something goes awry in the editing process*

To make a backup, make a copy of any audio files that you will be using, and change the file name to specify that the two files are the edited and unedited versions. This means that no matter what the editing program does to your file, you have the original audio to go back to. This is particularly important if you are editing in Audacity, because some processes in Audacity make permanent changes to the file you're using.

If you're both recording and editing in Audacity, you can export your unedited files as 24-bit wav files to create a backup between the recording and editing stages. If there are multiple audio tracks in your session, you will need to solo the track you are exporting. Solo will turn off every track except for the track you have chosen. This function is available underneath the track name to the left of every audio track.

Figure 8-3. *The solo function appears underneath the track name to the left of every track in Audacity*

To export your files in Audacity, go to File ➤ Export ➤ Export as WAV as in Figure 8-4.

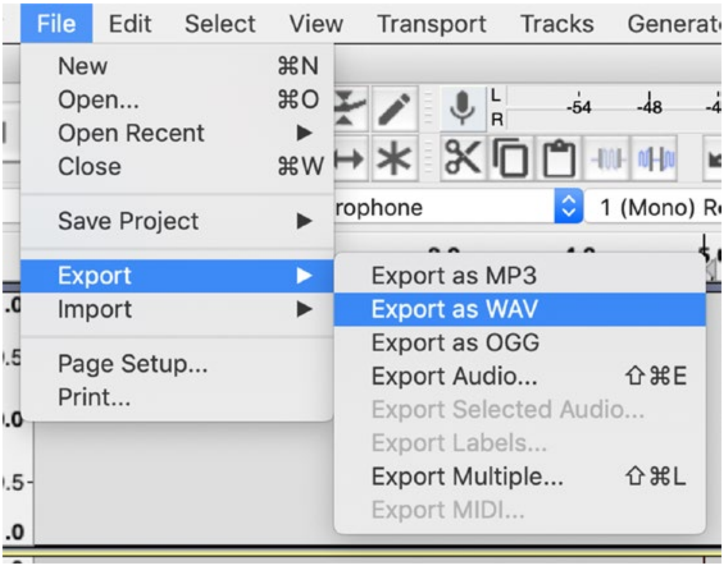

Figure 8-4. *Exporting a file in the wav format in Audacity*

A dialog box will appear which will enable you to choose where to save your files and to choose the bit rate at which to save them. There are a number of options available, and "WAV (Microsoft) signed 24-bit PCM" (as depicted in Figure 8-5) is the correct one.

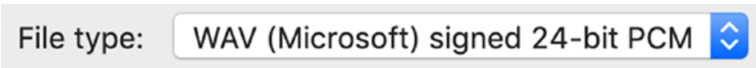

Figure 8-5. *Choose your bit rate*

Saving backups of your original files will save you from losing valuable audio data.

All Killer, No Filler

One of the great things about creating your own podcast is that it doesn't have to be any particular length, so you never have to cut out any great content because you've run out of time. This can also be one of the worst things about podcasts. There are too many podcasts in the world that contain one long, rambling, unedited interview. Don't be that guy. A long, unedited interview can give the listener the feeling of picking through a large bowl of bran hoping for a few sultanas. As the creator of the podcast, you should be doing that work for them. Once you have collected your audio, mark up the best content in your editing software, and then create a document where you organize the content into an interesting story.

Figure 8-6. *Listening to a long, rambling, unedited interview is like picking through a large bowl of bran hoping for a few sultanas*

When making a podcast, the process of collating and organizing information is ongoing. There are many parallels between fiction and nonfiction work. A good nonfiction podcast will start with research, and a good fiction podcast will start with scripting. Work develops. Ideas are refined. It's good to present only the best content. In both fiction and nonfiction podcasts, it's necessary to make notes and keep organized.

I have already touched upon the necessity of keeping notes when you're collecting audio in the field. You should also keep notes on any interviews recorded at home or in a studio in a document where you have collected all of your research.

At the 2020 Audiocraft podcasting conference, Wendy Zukerman from *Science Vs* took us through her process. She said that after every interview, she asks herself "what's that piece of tape that I'm going to run into my friend and be like, 'Oh my God, she said this,' like, what are those bits of tape? And then can I put those bits of tape in a story already?"[1]

My process is similar. After I have conducted an interview, I listen back to the audio and mark the bits that I thought were good and why.

Most audio editing programs, including Audacity, have a function where you can add markers (see Figure 8-7). To create a marker in Audacity, start by listening to your audio. When someone says something interesting, move the cursor to where you think the start of the statement was and click. Then press Cmd+B on a Mac or Ctrl+B on a PC. I think of it as "B for bookmark." A label will come up. Type in why you thought that piece of audio was interesting, and press Enter. The audio will just keep playing while you do this, and you can repeat this process as many times as you like.

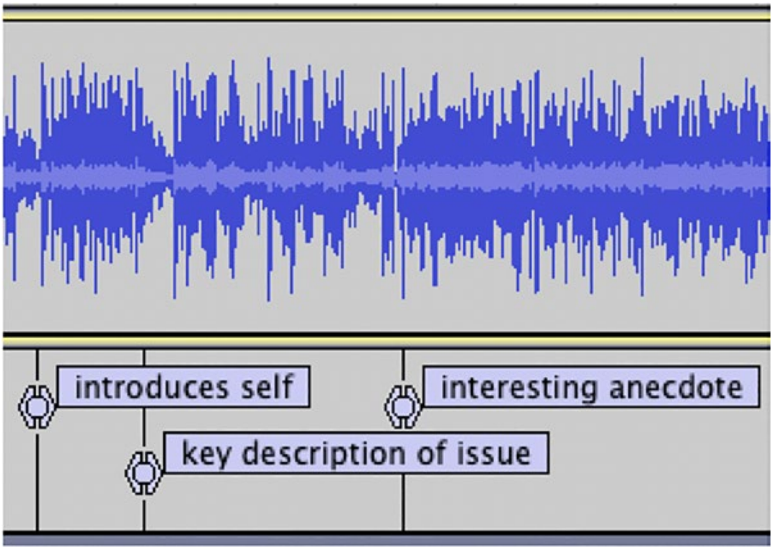

Figure 8-7. *Marking audio with labels (Audacity software)*

[1] Audiocraft, 2020. *Audio Journalism Vs. Covid-19.* [podcast] Audiocraft Podcast. Available at www.audiocraft.com.au/audiocraft-podcast-season-5-ep-1

When I'm labeling tracks, I just mark up what I think is good and leave the rest. There's no need to get precious. No one will ever miss content that's almost good enough, and most interviewees will appreciate you editing together their most interesting content.

If you are recording a fiction podcast, you can use the same process to mark which were the good takes.

If you want to edit or delete a label in Audacity, then right-click the label to find these options.

You can move labels from left to right in Audacity by clicking and dragging the circle in the middle. You can also extend them to mark the beginning and end of a piece of content by dragging out the arrows on either side of the circle.

You are now moving from the research phase into the storytelling phase, and you will need a new document.

Creating an Outline Document

The best podcasters I have worked with will send me a document where they have organized their podcast into a story based off label tracks. For more complicated material, I usually work with a full transcript.

Most audio editing programs will allow you to export the labels you have made into a text document. In Audacity, you do this by going to File ➤ Export ➤ Export Labels..... Unfortunately, Audacity exports these labels with a time reference that is in seconds and milliseconds, which is information that would only be useful to a robot. If you're a human, you can convert these time references into minutes and seconds using a spreadsheet or manually enter the correct time value next to the labels in your document. This is one of the many compromises you will face when using a free program.

Whether you are working with the labels or a transcript, use your outline document to map out how you're going to fit the pieces of audio together. For example, you can group audio by topic and splice together different guests' opinions. Or you could group content chronologically with different guests contributing to different parts of the story. Making a document sounds laborious, but ultimately it saves time and it will improve the quality of your work. The document will also help you script your voice-over.

The only time I wouldn't use an outline document is if I were editing a podcast that consists of a single interview that has been conducted by a talented interviewer with an interesting guest. In this case, I would mark up the good parts of the interview, and then I would simply delete the rest.

To delete audio in Audacity, make a selection, and then press the delete key.

You will piece together your audio clips using the outline document as a guide.

Piecing Together Your Audio

Now that you have created your outline document, you can start a new session in your editing program and put all of the pieces of audio in the right order. You will want to have a different track for every source of audio.

Having different tracks for every source of audio will make the editing process a lot easier. There should be one track for every person speaking, and there should be individual tracks for every other type of audio. If you have different songs in your podcast, you will want them on separate tracks. This will help in the balancing and mixing of your work. You might decide that a single speaker should be slightly louder to fit in with others. If that speaker is on a separate track, then changing their overall level is as simple as moving the fader. If they're on a track with other sources of audio, then you will have to change every one of their pieces of audio individually. Similarly, if you are adding any kind of processing such as equalization, you will want to individualize it to a single source of content on its own track. In most audio editing programs, it's also possible to add processing to the master fader, which will then affect all of the audio.

To create a track in Audacity, go to Tracks ➤ Add New, and choose either a mono track or a stereo track. Alternatively, you can drag and drop an audio file straight into Audacity from Windows Explorer or Finder. If you are dragging and dropping, be aware that Audacity changes the sample rate of the whole session to that of the first audio clip that you add. I cover this topic in Chapter 1, "File Formats and Settings," in the subsection "Setting the Bit Rate and Sample Rate in Audacity." When I'm editing, I like to have two tracks for each source of audio – one for the original file and another track where I am adding the sections of audio that I am taking from the whole.

To help you stay organized, each of your tracks should be individually named. To change the name of a track in Audacity, click the arrow to the right of the track name. A menu will appear. Select "Name..." as in Figure 8-8.

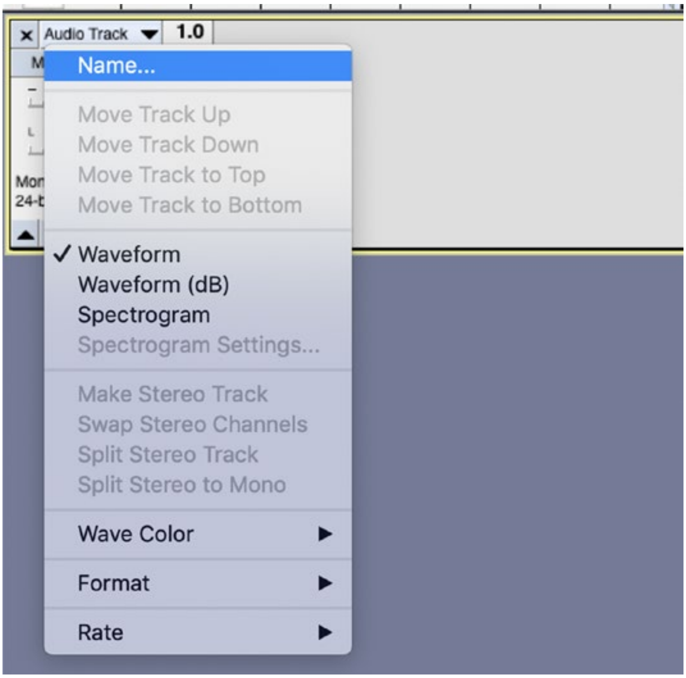

Figure 8-8. *Changing the name of a track*

If you have multiple tracks in Audacity, you will often want to mute one or solo one. Mute will turn off one track. As previously mentioned, solo will turn off every track except for the track you have chosen. These functions are available underneath the track name to the left of every audio track.

Figure 8-9. *The mute and solo functions appear underneath the track name to the left of every track*

Now that your files are in Audacity, you can cut and paste them in a similar way to how you would cut and paste text in a document. Select the segment of audio that you want, and then on a PC, use Ctrl+C to copy or Ctrl+X to cut. Select the place that you want to paste the audio with your cursor, and press Ctrl+V. If you're using a Mac, that's Cmd+C to copy, Cmd+X to cut, and Cmd+V to paste. You can use Ctrl+I (PC) or Cmd+I (Mac) to make a break in the audio. I think of that as "I for incision."

You might want to drag pieces of audio around, but before I explain how to do that, I need to digress. Audacity has six options for its cursor as in Figure 8-10.

Figure 8-10. *Audacity's cursor options*

The buttons in Figure 8-10 are

Selection tool	Envelope tool	Draw tool
Zoom tool	Time shift tool	Multi-tool

You can choose the different cursor options using the buttons depicted in Figure 8-10. These can be found on the main screen above the audio tracks. It is quicker to control the cursor options using the function keys on your keyboard:

F1 – Selection tool. This is the default and is used to select audio.

F2 – Envelope tool. This is for changing the volume of a piece of audio.

F3 – Draw tool. This lets you literally draw a sound wave which is not that useful for podcasting.

F4 – Zoom tool. Left-click for zoom in, right-click for zoom out.

F5 – Time shift tool. This is the tool that you need to drag pieces of audio around.

F6 – Multi-tool. This tool is a combination of all of the tools so far mentioned. In multi-tool mode:

- The selection and envelope tools will appear depending on where you have moved your cursor.
- On a PC, you will need to right-click to zoom out. If you right-click and drag to make a selection, then Audacity will zoom to your selection. On a Mac, you will need to

hold down Ctrl and click to zoom out and hold down Ctrl while making a selection to zoom in.

- If you want to move audio around while using the multi-tool on a PC, you will need to first press Ctrl. On a Mac, you would press Cmd to move the audio.

To navigate around Audacity, you can scroll horizontally or vertically. Press "Home" on your keyboard to go back to the beginning of an audio clip and "End" to go to the end.

Ctrl+A (PC) or Cmd+A (Mac) will select all of your audio. Double-clicking any audio clip will select all of that clip. A single click with the selection tool will unselect audio.

The spacebar starts and stops audio at the selection.

By cutting, pasting, deleting, and moving audio, you will have created the bare bones of your podcast. Have a listen to your work so far. Now that you can hear the audio together, you might want to refine the order. You might need to go back and get some linking content from your original recording to improve storytelling. Alternatively, you can explain the links between your clips with voice-over. Think about pacing; a voice-over can concisely move a story forward, whereas a section of an interview might draw attention to something important. You might set the scene with voice-over and then dwell on details such as a description of a person's emotional state or an interesting anecdote. For example, in an episode of *Let's Talk About Sects* that I recently edited, the guest described leaving a cult as "like leaping out of a plane into the pitch black." This kind of powerful imagery leaves a lasting impression. I might even use a snippet like this as a short teaser in the introduction to pique the listener's interest. When it comes to writing the voice-over, you can script it in the outline document you created earlier.

It's important not to misrepresent what people are saying, so even though you might choose to include only a small portion of what you record, ask yourself if what you have chosen represents the interviewee's overall views. Sometimes I am tempted to simplify a complicated matter to make it more audience friendly, but there is always a way to explain complicated material for a general audience. Ultimately, these complications make for better storytelling.

Once you have finalized the audio that you're using, record the voice-over and add it to the session in its own track.

The outline document you created to organize your podcast can now be turned into a transcript. There are a number of benefits to publishing a transcript of your podcast. A transcript makes a podcast accessible for a much wider range of people including those who are hearing impaired, people with an auditory processing disorder, people who are autistic, or people whose primary language is not the one you are podcasting in.[2] Erin Kyan[2] from **Love and Luck** goes further and creates a video version of his podcast that combines the audio with captions for the purposes of accessibility. Another benefit of a transcript is that it enables a person who is looking for a piece of information to quickly find it without having to listen to your whole show. A transcript will also help people locate your podcast through search engines more effectively.

Now that you have the main structure of your podcast, it's time to start thinking about cleaning up your audio.

You *Can* Edit It, but Should You?

As well as being the former prime minister of the UK, Winston Churchill was an occasional broadcaster at the BBC. He was a well-known orator, but he suffered from both a stammer and a lisp. While he made efforts to overcome his stammer and his lisp, they would occasionally come through. He realized the value of his unique voice when he wrote

[2] Audiocraft, 2018. *Under the Hood with Love and Luck*. [podcast] Audiocraft Podcast. Available at: https://play.acast.com/s/audiocraft/underthehoodwithloveandluck

“Sometimes a slight and not unpleasing stammer or impediment has been of some assistance in securing the attention of the audience.” After Churchill overcame his lisp, he made efforts to get it back, because it was part of what made his voice recognizable. He even had specialist dentures made that preserved his lisp.[3]

Figure 8-11. *Churchill had special dentures made to preserve his lisp because he realized it was part of what made his voice unique*

It would be possible for a sound engineer to change Churchill’s speech pattern to make it more conventional, but at this point in history it would be sacrilege. A person’s voice and accent is unique. It reflects their history, their culture, and which groups they identify with. Many people think that there’s a right and wrong way to speak. For too long, it was accepted that a “good voice for radio” was that of an upper class, educated, able-bodied, white man in the city. Should we presume then that everyone else has a bad voice for radio? As another example, you can easily change the accent of a regional Australian to make it sound less nasal. To some

[3] G, J., 2012. *Winston Churchill’s Dentures*. [online] Speech Buddies Blog: Speech, Language & Pronunciation Guides. Available at: www.speechbuddy.com/blog/speech-disorders/winston-churchills-dentures/

people a nasal accent is unpleasant, but to me it sounds like home. One of the best things about radio, and in particular podcasting, is that due to its relatively low costs it enables a wide variety of people to put their ideas out into the world. I would only change a person's speech patterns during the interview segment if it's particularly distracting and is detracting from the content.

And so when it comes to editing an interview, I am a minimalist. When I've found the audio clips that I want, I tend to spend very little time taking out quirks of speech. I partly do this because it's quicker, but mostly because I want to capture the unique essence of a variety of voices.

It will be tempting to want to change the sound of your own voice in the editing process. When you first listen back to a recording of your own voice, you will discover that it sounds horribly wrong. This is because you're used to hearing your voice from within your head, not from outside your head. The slight difference is like a parody. It's as if you're back in school and a classmate has affected a cruel imitation of your voice to repeat back everything you've said. Everyone is laughing at you. This unfortunate phenomenon is the same for everyone. You don't have a weird voice, and you don't have to fix it. You will eventually get used to the sound.

In the case of a fiction podcast, you might want to change the character of a person's voice for creative reasons, such as if the voice actor was playing a talking sandwich.

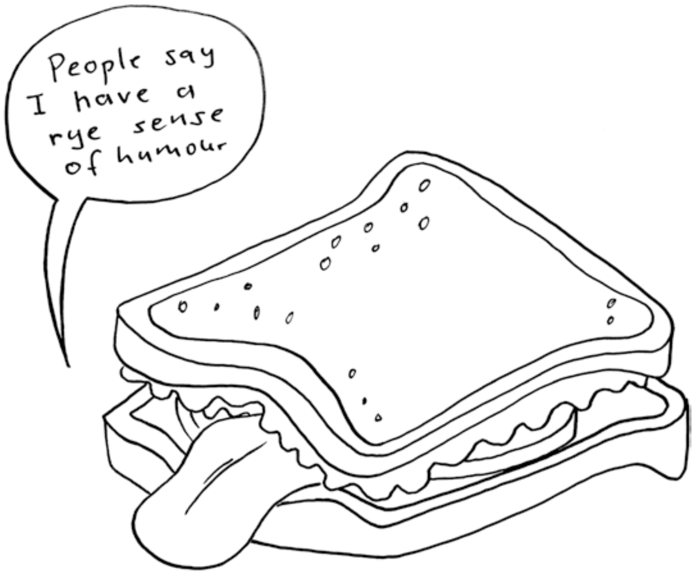

Figure 8-12. *Changing a person's voice so that they sound like a talking sandwich would be an interesting creative challenge*

You can make a person sound less hesitant by cutting out their umms, aahs, repeated words, and pauses. This is relatively straightforward, but ask yourself if you're losing meaning. Perhaps there is a reason that the person is hesitant. Are you helping the person put their best foot forward, or are you changing the tone of what they're saying? I once asked a woman a very difficult question that she didn't want to answer. At first, she dodged the question with some irrelevant information, and then she answered it. It was a sensitive issue and answering it might have had ramifications for her. I initially edited the clip with my original question and her final answer, but I felt uncomfortable about it. I went back and re-edited it so that the listener could hear the entire question and answer, including the long section of her stalling and dallying. The whole clip represented her views better and ultimately made for more powerful storytelling.

Talking is not the only way that humans communicate. A moment of silence might be a good place to cut, but once again, you should consider meaning. Silence can be wonderfully expressive.

Figure 8-13. *A silence can be expressive*

When placing audio clips, I'm always making decisions around the length of silences. A longer gap between audio clips can give a story a sense of calm, or it can express the seriousness of the content. A really long gap gives the listener room to think about an important point. Short gaps between audio clips pick up the pace and add excitement.

The quality of a person's breathing can be expressive. It might pay to leave in a little sigh at the end of an audio clip or a sharp intake of breath at the beginning. Some people take a particular breath that means "this is the beginning of a new topic," and it seems a shame to edit that out.

There are a few circumstances in which I might remove the sound of breath in a voice-over. A voice-over that is devoid of breathing can sound clean and professional. This style of editing a voice-over was more commonly used in the past and can still be found in certain fields such as advertising. You might also want to remove sounds of breathing if the person giving the voice-over has a sniffly nose.

You can strike a compromise by cleaning up audio for the sake of the audience, but also leaving in some of the person's unique speech patterns. I once edited an interview that was conducted with a deaf person using an interpreter. There were large pauses in the speech to accommodate this, which made me think that listeners would wonder whether the audio had stopped. To remedy this, I cut down the length of the pauses, but I still left some of the pause for effect. Similarly, I sometimes come across someone with a speech habit that I find is distracting from the content. An example would be the overuse of a phrase like "do you know what I mean?". I will leave a few in and take a few out.

An exception to my minimalist editing is the voice-over. A clean, well-recorded voice-over sounds authoritative and binds the story together. It can help keep the audience's attention even if some of the other audio is lacking. A fiction podcast is another example where you would take the time to edit out any mistakes. My main editing advice here is to record and re-record this sort of audio until you've got it right.

While I advocate for minimalist editing, some adjustments will need to be made so that the audio sounds consistent.

Managing the Loudness of Your Podcast

Most people will remember the bad old days when advertising on television and radio was much louder than the program material. At this time, there were standards as to how loud audio could be at its peak. Advertisers would get around these standards by *compressing* their audio. To compress is to reduce the dynamic range, or the difference between the loudest and the quietest sounds. So while a drama may have had noticeable differences between the level of a shout and a whisper, an advertisement would consistently be at shouting level. The effect was quite jarring. You'd be getting all tearful over a scene where two lovers were pledging that they would always be together and then all of a sudden it was "***RUGS!!***"

It sounded a bit like Audio 8-1.

Audio 8-1 The loudness of program material compared to an advertisement

This shows that there are two ways to consider loudness. There's the true peak, which is the level of the loudest bit of audio within a piece. There's also how loud an overall piece of audio sounds to the listener, which is described in audio terms as just *loudness*.

If you look at the waveform in Figure 8-14, which is a visual depiction of the two sound files you just heard, you can see the difference in loudness between these two types of material.

Figure 8-14. *The waveforms of the program material (left) and the advertisement (right) from Audio 8-1. The true peak of the program material is circled in red*

The waveform of the program material on the left has some quiet bits and some loud bits, whereas the waveform of the advertisement on the right is consistently loud.

If you look closely at the bottom left-hand side of the program material, you will see that the sound wave peaks at the same level as the advertisement. Even though the peak of these two files is the same, the loudness is different.

One way of describing this is that the audio for the advertisement is more *compressed* than the audio for the program material. Earlier in the book, I used the term *compression* to mean taking data out of an audio file to make it smaller. It means both things.

Figure 8-15. *TV used to have jarring changes in loudness, because the volume of audio was restricted based on the true peak, not the loudness*

Nowadays, sound engineers have ways to measure loudness, instead of just the peak. As such, material that is broadcast on TV, radio, and online has to meet loudness standards. When you're listening to Spotify, you don't have to change the volume between tracks because the program measures the loudness of the whole track and does it for you.

The algorithms that measure and match the loudness between different sources of audio are good, but they are not well suited to changing the loudness within a single audio file. Podcasts are notorious for rapid and inconsistent changes in loudness. It is up to individual podcasters to manage loudness within their own work.

Loudness in Context

A computer program can measure loudness using an algorithm, but you have something much more sophisticated: your ears and your brain. When editing your podcast, set the loudness so that your audio files have a consistent volume.

In the case of my example from the bad old days of television, you would need to turn down the ad to match the loudness of the program material. It would now sound like Audio 8-2.

Audio 8-2 Matching the loudness between program material and advertisement

Figure 8-16 is the visual depiction of those two audio files after I have matched the loudness.

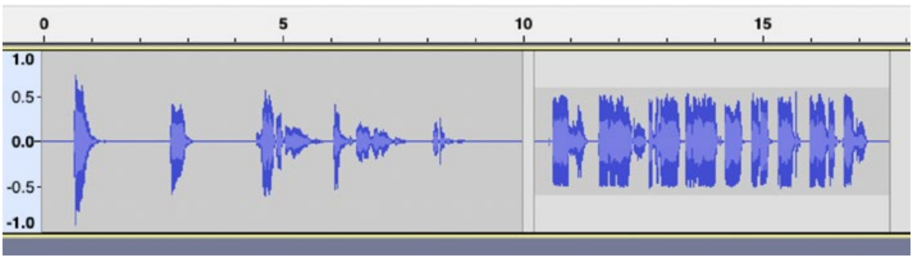

Figure 8-16. *Matching loudness levels between two audio files*

On the left is the program material which is the same level as before. On the right is the ad, which I have reduced in level to more closely match the program material.

When I was matching those two pieces of audio, I was doing something a computer couldn't do: I was thinking about context. The program material ended in a whisper, while the ad was all shouting. So, I didn't set them to exactly the same loudness, I made the ad a little louder than the drama. The difference between the shout and the whisper is still much smaller than it would be in real life.

A simple way to keep loudness consistent over a long podcast is to use the loudness of talking as a reference for everything else. Choose a particular value on your meter for audio of a person talking, and then set the loudness of everything else around that. A whisper would be quieter, a shout louder. Maybe you would run a piece of music quietly underneath your voice-over, and the sound effect of a fire truck would be louder than the shouting.

You can balance speech, music, and sound effects within a pretty limited dynamic range and still give the impression of changes in volume because the peak of an audio file is not the only thing that makes it sound loud. You can see in Figure 8-16 that the peaks in the audio file on the right are now lower than the peaks in the audio file on the left, yet the file on the right sounds louder. This is because another important element in loudness is the duration of the peaks. In the audio file on the left, the peaks quickly drop off, while in the audio file on the right, they stay consistent for longer. You can also take advantage of your listeners' imaginations. Something like a shout sounds loud because shouting changes the character of the voice, so it doesn't necessarily have to be that much louder.

Managing Loudness in Audacity

There are a few different ways to change the level of an audio clip in Audacity. You can use volume envelopes, change the gain, or use the mixer board.

Volume envelopes are good because they give you a visual representation of loudness, although this guide shouldn't replace your ears.

To use a volume envelope, press F2. This will change the look of your audio tracks. A purple line will appear at the top and bottom of the track, and there will now be two shades of gray instead of one gray area.

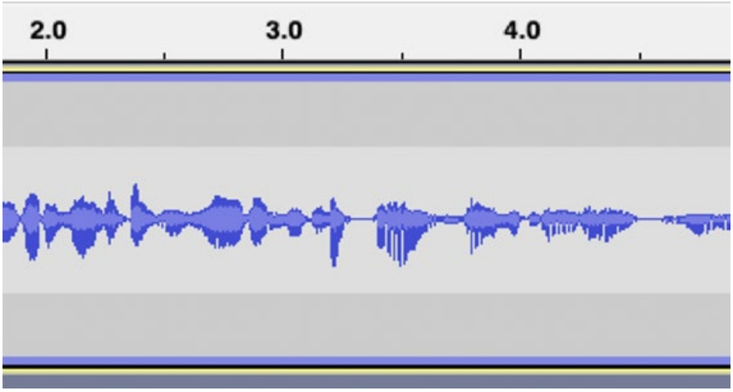

Figure 8-17. *The appearance of an audio track after the envelope tool has been selected*

If you click the purple line, then four white dots will appear. Move any of those dots to change the level of the entire piece of audio consistently.

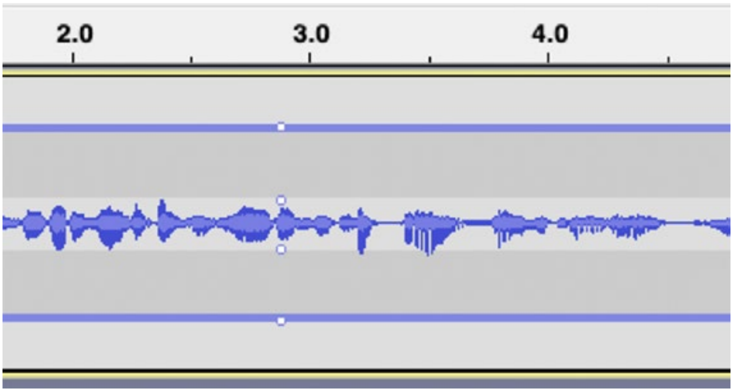

Figure 8-18. *Moving the white dots will change the level of the entire piece of audio*

If you click in an additional place on the purple line, then more white dots will appear. Now you can shape the volume envelope as you wish. You can add as many white dots as you like.

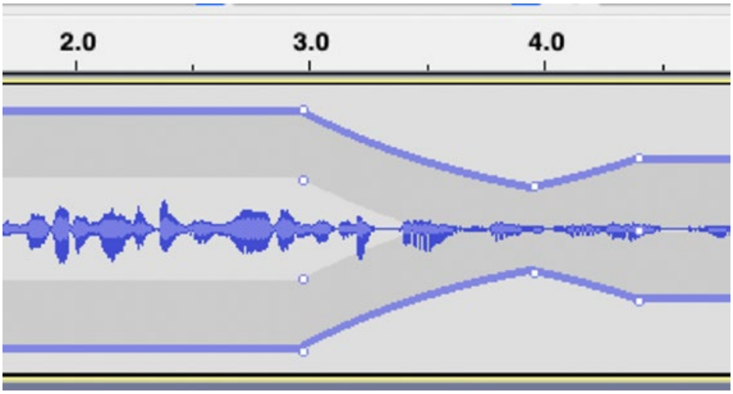

Figure 8-19. *Add more white dots to shape the volume of an audio clip*

If you want to delete a white dot, drag it off the top or bottom of the track.

Another way to change the level of your audio in Audacity is by using the gain function. This can be found to the left of the audio track underneath the name of the track.

Figure 8-20. *The gain function in Audacity*

Your third option for changing the volume level in Audacity is to use the mixer board. You can find this by going to the file menu and choosing View ➤ Mixer Board.... A window will come up as in Figure 8-21.

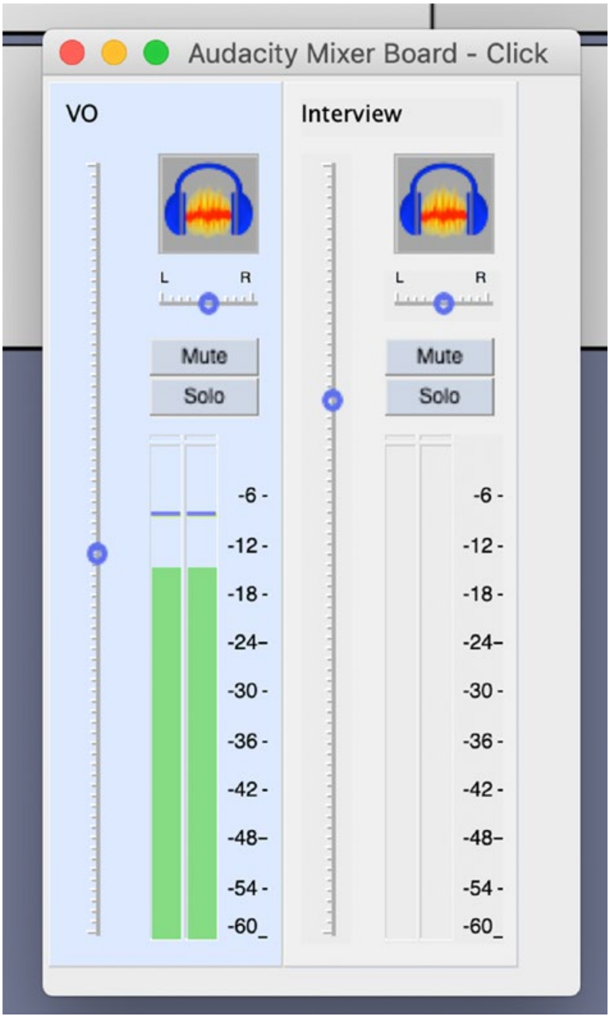

Figure 8-21. *The mixer board*

This method of mixing is good because you can see metering for individual tracks.

These are not the only meters you should be watching. When you're combining more than one track, the total volume might exceed the peak. Make sure to keep an eye on the master meter which is at the top of the main screen and looks like Figure 8-22.

Figure 8-22. *The Audacity master meter*

If the meter has exceeded the peak, it will look like Figure 8-23, and you will need to turn some of your clips down.

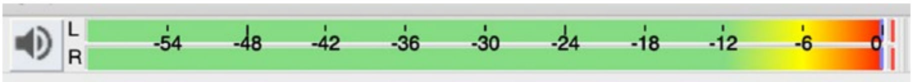

Figure 8-23. *The Audacity master meter when it is peaking*

You might not have seen your meter clipping, but Audacity has you covered. If you notice that a red line has appeared to the right of your meter, it means that your audio peaked at some point.

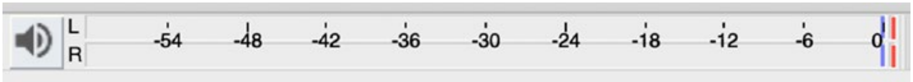

Figure 8-24. *The Audacity meter is telling you that the audio has peaked at some point*

Once you have noticed this red line, you can clear it by clicking it. Then you will have to go through your audio to find the place where it is peaking and resolve it.

When you are in the process of creating a consistent loudness over the entirety of your podcast, you might run into the problem of a piece of audio that's very quiet, but has a really loud part that prevents you from turning it up to match the rest. The solution to this is compression.

Understanding Compression

Compression is a process whereby the loudest parts of an audio clip are automatically turned down, so as to reduce the difference between the loud and soft parts of that piece of audio. Compression is useful for helping you level out the dynamic range of your audio and balance the different parts without any files peaking. Using a compressor is hard to get the hang of and is therefore beyond the scope of this book. However, compression is still an important concept to understand. I will discuss a simple version of compression that is manual rather than automatic.

Most audio that you hear is compressed for both artistic and practical reasons. Compression helps sound engineers balance different sources of audio. When a song is first recorded, the bassist might play some notes loudly and some notes softly. The sound engineer might then compress the bass so that it is consistently louder than the guitar. Further compression will be added to the whole song for artistic reasons and to meet the requirements of different storage media such as vinyl. If you've ever thought that a song sounds different when it's played on the radio, then you would be correct – it is much more compressed than the CD version. One of the reasons for this is that there's an expectation that many people will listen to radio in a loud environment such as their car where any quiet parts of a song might be missed. If the difference between the quiet parts and the loud parts of a song are reduced, then you can hear the whole song in peak-hour traffic.

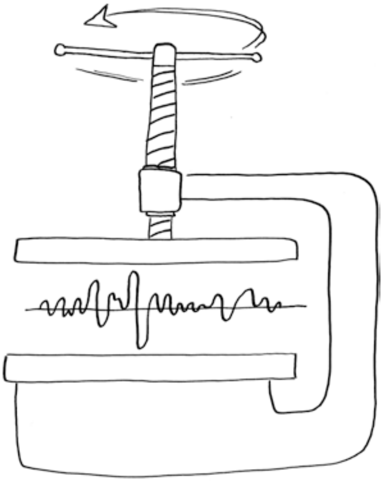

Figure 8-25. *You can think of compression as a process which squashes down the peaks of audio, thereby reducing its dynamic range*

Applying Compression Manually

In terms of podcasting, you might want to use compression in a situation such as in Figure 8-26. In this example, someone is talking at a particular level and there's a sudden loud sound.

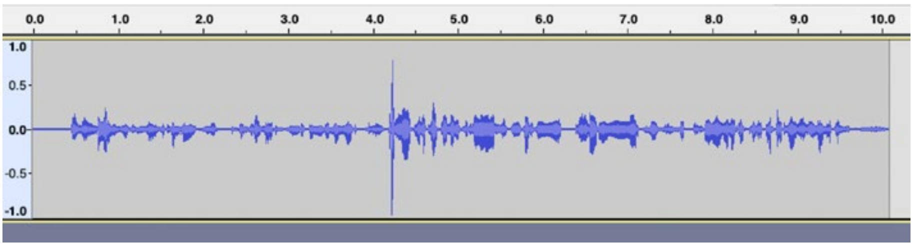

Figure 8-26. *Audio file of person talking with a sudden, loud sound*

The person might have hit the microphone stand with their hand, or thumped the desk, or made a click noise with their mouth. You could set a compressor to automatically turn down that sound to the level of the rest of the audio. It's difficult to use a compressor skillfully, so the easy but laborious solution is to manually separate out the loud audio from the rest of the audio and then turn the loud bit down.

In Audacity, you would press F4 to access the zoom tool. Zoom into that section of audio. Then press F1 to go back to the selection tool. Make sure that you're just selecting the loud sound and not any bits of silence that are on either side of it.

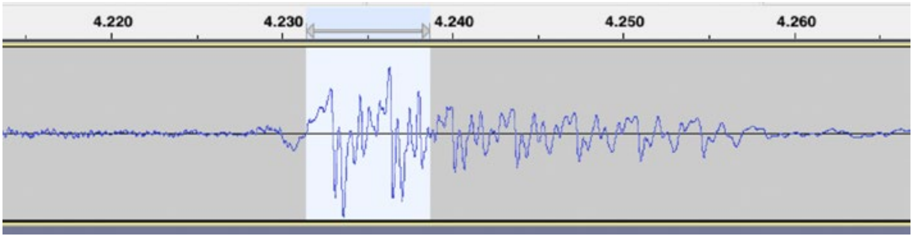

Figure 8-27. *Select only the loud part of the sound, even if it is a small portion of a whole word*

Ctrl+I (on a PC) or Cmd+I (on a Mac) will make two cuts to separate out this section.

Press F2 to choose the envelope tool. A purple line will appear at the top and bottom of your audio. If you click this line, a white dot will appear. Drag the white dot down until the sudden, loud sound is about the same height as the surrounding audio as in Figure 8-28.

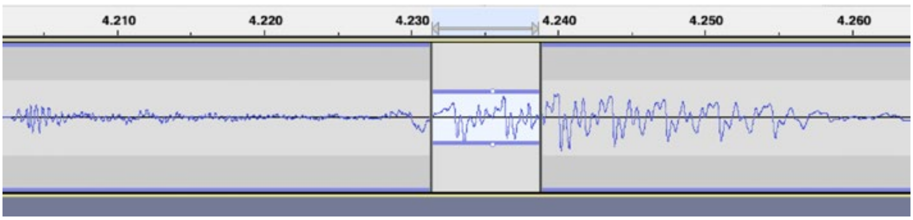

Figure 8-28. *Use the envelope tool to turn down a sudden, loud sound*

Now zoom out (F4), go back to the selection tool (F1), and listen to your audio.

If the loud sound is unwanted, another option is just to delete it. Once you have turned down or removed the loud sound, you can turn up the entirety of that audio file using the mixer board, so that it better fits within your podcast.

In the last chapter, I discussed a situation where a person was using a single microphone to interview multiple people. I included an example of an audio file where three people are talking at different levels. I've repeated the example in Figure 8-29.

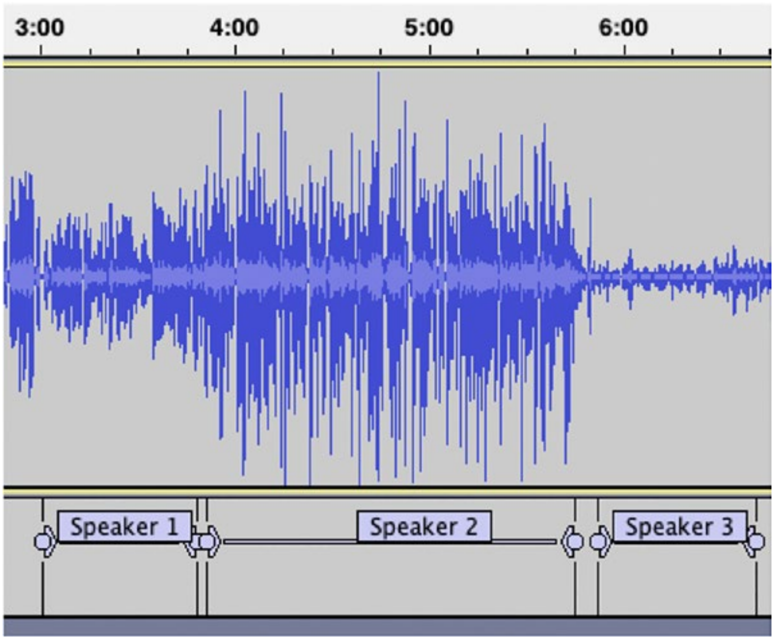

Figure 8-29. *Audio of three different people talking at different levels*

In this example, you have three different people talking on one track. The first person's levels are good. The second person's levels are too loud and they're peaking. The third person's levels are too soft. It's better to avoid making recordings like this, but everyone makes mistakes.

A sound engineer would use compression and a few other tricks to even out this audio, but that is a bit complicated. The simple but laborious solution is to manually fade the volume of this track up and down using volume envelopes. In the last example, we separated out the loud section and changed the volume, but that doesn't work with these longer clips. As you change the volume of the speech, you're also changing the level of the background noise. This is much less noticeable if you use a fade.

Use the envelope tool to even out the level of the audio as in Figure 8-30. If you look closely, you will notice that Figure 8-30 is the same as 8-29, except that I've cut out large sections so as to fit it onto the page.

So, in Figure 8-30, I have labeled the three speakers. I turned speakers 1 and 2 down, and I turned speaker 3 up. I have labeled a section of the audio as "agree." This is where speakers 1 and 2 briefly jump in to agree with speaker 3. In the "agree" section, I have briefly turned the level down and then up again. Now that I have leveled out the audio, you will notice that the waveform looks fairly even. More importantly, it sounds about the same.

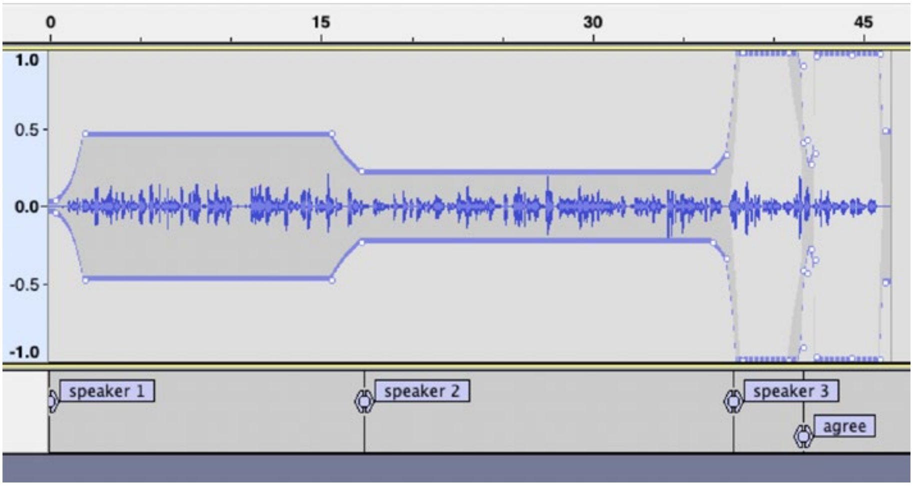

***Figure 8-30.** Using the envelope tool to even out the loudness of recorded speech*

This DIY compression can help you level out the dynamic range of your audio and fit everything together.

Compression in Streaming Services

The final thing I want to say about compression is that when someone plays your podcast through a streaming service such as Apple Podcasts or Spotify, these programs automatically add another layer of compression. If you were thinking earlier that it would be good if a podcast was compressed so that it could be heard in a loud environment such as a car, then you will be pleased to hear that Apple Podcasts and Spotify have you covered. These streaming services will compress your podcast, and they will change the overall loudness of your podcast to their standard. However, you will still need to even out the dynamic range *within* your podcast. These streaming services can reduce the difference between the loudest and softest parts of your podcast, but they don't fix inconsistent changes in loudness.

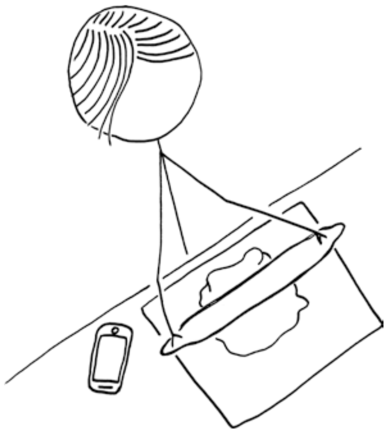

Figure 8-31. *Streaming services such as Apple Podcasts and Spotify add compression to your podcast so it can be heard in a noisy environment, such as a kitchen*

Applying compression to your podcast is essential to balancing different elements and to creating an audio file that can be consistently heard in a loud environment. People often confuse compression with normalization, so I will briefly discuss the difference.

Normalization

Normalization is a process that turns up the level of a piece of audio until the peak reaches a certain value. Unlike compression, it does not reduce the difference between the loudest and softest parts of an audio file, it simply turns everything up or down. If you wish to normalize your podcast, I would recommend that you set the value to -0.5 dBFS.

As I mentioned in the last section, streaming services such as Apple Podcasts and Spotify will automatically bring your audio up to a specified loudness. If you think that your audience will be listening through these services, then normalization might not be crucial. Normalization is most useful in situations where people are listening to your podcast without that extra layer of compression, such as if they are listening through your website. Even if you're using a streaming service, normalization can still add value to your podcast by maximizing the amount of quality audio data in your file. If you're not using a streaming service, normalization will bring the final file closer to a level that will be useful for listening devices.

A mistake that some people make with normalization is to apply it to all of the separate audio clips within their session in their audio editing software. This will mean that the peak of every one of their audio clips is the same, but the loudness will be inconsistent. It will result in a file such as the one I used at the beginning of this section where I was reminiscing over the bad old days of television. When normalizing your podcast, do it after you have bounced the entire podcast into one wav file.

If you are normalizing your podcast in Audacity, bounce your entire session to a wav file. Import the new file into a new session. Select the audio by double-clicking the waveform. Go to the menu and choose Effect ➤ Normalize…, and a window such as the one in Figure 8-32 will appear.

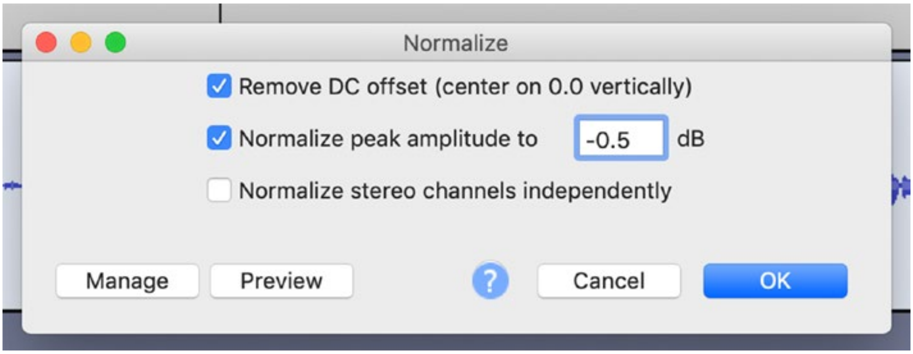

Figure 8-32. *Normalizing a track (Audacity software)*

Fill in the options as shown, and click OK.

This will turn up your entire audio file so that the peak is at -0.5 dB. You can now export that file as a wav or as an mp3.

Fades

So you have all of your pieces of audio lined up and at about the right loudness. The next step is fading every audio clip in and out to avoid jarring cuts to silence and those little clicks you'll sometimes hear when you cut a piece of audio. Fades will also help you fit pieces of audio together in a pleasing manner.

Every audio clip that you use in your podcast should be faded in and out again. You might not think this is necessary for a recording made in a silent room, but "silence" is relative. A recording of a "silent" room is called *room tone*. It will be ever so slightly louder than digital silence, which is the silence you will find in the gap between your audio clips when you play them in your editing program. Even if something is recorded in what sounds like a silent room, it is quite noticeable if that almost imperceptible amount of sound is suddenly gone. Hearing a sudden cut to silence gives me the same feeling as when I'm walking down stairs and I miss a step –

something I subconsciously thought would be there is now gone. This is why you should fade all of your clips in and out to digital silence.

Figure 8-33. *A sudden cut to digital silence can give the listener the unnerving feeling that something they subconsciously expected to be there is now gone – a bit like missing a stair*

The length of a fade is important. A clip that only has room tone might need a relatively short fade. A clip with a lot of background noise will need a longer fade to sound smooth.

There are exceptions to the need to fade clips in and out. If two pieces of audio have the same room tone and are run together, then you do not need a fade. For example, if a person is talking and you want to delete the middle of their sentence, then you don't need to fade out the beginning of the sentence and fade in the end.

Sometimes I use fades to subtly communicate to the listener that I am putting together two pieces of audio that were not from the same part of the conversation. A person might have spoken about the same topic twice at two different times in an interview, and now I want to put those clips together. It might be possible to combine them without a fade, but I would still fade out the first clip and fade in the second because it feels more honest to me.

Sometimes a piece of audio needs to be cut off very quickly, without room for an adequate fade. You might be interviewing a person who changes topics but never finishes their sentence which is a habit that people develop as a defense against being interrupted but is not that suitable in the context of interviews at least when they're being conducted for a podcast, and so on. If you are editing such an interview, make the cut straight after a word finishes, then grab the next piece of silence (as in room tone), place that immediately after your cut, and then fade the room tone out.

Another strategy that people use to smooth out their editing is to run a track of room tone underneath their whole podcast. This can smooth out cuts, quick fades, and rough edits. To create a room tone track, make a recording of a silent room. Alternatively, you can download one of these tracks from the Internet, although you might have to consider copyright.

A cross fade is used to transition smoothly from one piece of audio to the next. You start with two pieces of audio on different tracks, and you fade out one as you fade in the other as in Figure 8-34. In this example, I'm fading out a song and then fading in the voice-over.

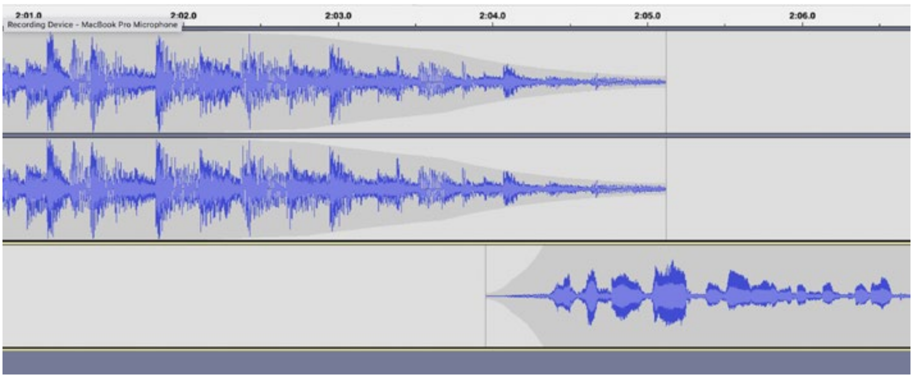

Figure 8-34. *A cross fade*

Fades help audio clips to flow and to fit together smoothly for a polished podcast. Fade all of your audio clips in and out to fit them together and to avoid clicks or a sudden drop to digital silence.

Summary

Podcasting may be a relatively new medium, but you can think of it as an updated version of the fundamental human need to communicate with those around us and express ourselves creatively. Podcasting is storytelling, and editing is the stage at which you bring together your best content and shape it into a story. The process gives you a chance to consider the arrangement of the elements and the pacing. Keep the story flowing by ensuring consistent loudness and by skillfully using fades.

In this chapter, I have really only touched upon the basics of editing. This is partly because it is easier and more effective to make a good recording than to fix a bad recording. Another reason that I have covered only the basics of editing is that understanding the audio effects and processing used by sound engineers is quite complicated and beyond the scope of this book. With the knowledge that you've gained from this book, you should now be able to engage with audio industry resources on topics such as editing. If you want an expert job done on your editing, then consider hiring a sound engineer to bring their skills and expertise to your podcast.

So far we've discussed how to use editing techniques to tell a story. In the next chapter, we'll heighten the story using music, atmos, and sound effects.

CHAPTER 9

Music, Atmos, and Sound Effects

I've recently started listening to the ABC radio podcast called **The Pineapple Project**. Season one is about managing your money – an important but dull topic that they've somehow managed to make cool. After so many years of ignoring my own mother's sensible advice on this topic, I have a newfound mania for savings accounts and insurance.

My enthusiasm is probably quite annoying to my mother. What does "Sensible Emily" from The Pineapple Project have that my mother doesn't? I'll tell you: a theme song.

The Pineapple Project uses a lot of strategies including music, atmos, and sound effects to bring listeners into a story, and you can too. These elements can add interest, help listeners place the story in a time or place, move the story along, and heighten emotional content.

C. M. Green, *The Podcaster's Audio Handbook*,
https://doi.org/10.1007/978-1-4842-7361-6_9

Figure 9-1. *Whacky sound effects might help your listener engage with a subject they find boring*

When using atmos, sound effects and music you should consider the needs of your audience. If you expect many members of your audience to be hard of hearing, then layering other elements with speech might obscure the intelligibility of the words for these listeners. In this case, you can place music and sound effects next to your recordings of speech. Generally speaking, you should prioritize speech intelligibility even if you decide to layer it with atmos, music, and sound effects.

Atmos

In the last chapter, when I was talking about loudness, I introduced you to a piece of program material that I paired with an advertisement. The clip featured two lovers who would always be together. When I recorded the dialogue, I imagined that they were in a restaurant, but what if they were in a spaceship? *Atmos* is the ambient sounds you'll hear in any location. I've added a layer of "spaceship" atmos, including machine hum and the sound of air filters, to my dialogue track. This moves the story to an exciting new location.

Audio 9-1 Lovers in space

Atmos can help a listener place a story in a physical space. It can be used to enhance outside recordings or to situate the dialogue of a fiction podcast in any location.

If you have conducted an interview outside and have also recorded some atmos, you can run it under the interview, and you can even extend it to run under the voice-over. You can use it to signal that the listener has entered a particular place before anyone starts talking, as a filmmaker would do by introducing the visual of a set.

You can use atmos to signal a theme or emotion. You might be making a fiction podcast set in a peaceful suburban street. If your character is stressed or overwhelmed at one point in the story, you could change the atmos to move them to a different, more stressful location such as a busy road. Or you could add a few sound effects that you might find in a suburban setting such as a lawn mower or a baby crying to enhance the feelings of stress.

Sound Effects

When we left our lovers, they were pledging their love in a spaceship. With the addition of sound effects including the insistent beep of machinery and the roar of the rockets, the spaceship is now launching them into an exciting interstellar adventure!

Audio 9-2 Lovers in space – Liftoff

Like atmos, sound effects can enhance storytelling in a fiction or nonfiction podcast. They can help place the listener in an environment, move the action along, or signal emotion. On the negative side, they have the potential to be distracting or off-putting. When using sound effects, consider whether they're adding to or subtracting from the storytelling.

The best way to get sound effects is to record them yourself. You can also get sound effects from websites such as **freesound.org**. Look for clips that have been recorded as a wav file. You will want files where the sounds have been recorded at a healthy level – that's somewhere between too high and too low. The following pictures (Figures 9-1 to 9-4) are examples of audio files from freesound.org. They are all recordings of dogs barking. You can assess the visual representation of the file to get a quick idea of whether it has been recorded well before listening to it.

You don't want a sound effect that has been recorded too quietly, such as in Figure 9-2.

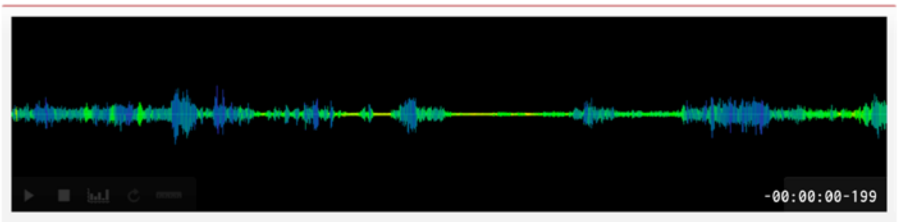

Figure 9-2. *An audio clip from freesound.org where the sound of the dog barking is barely louder than the background noise*

You also don't want a sound effect that's been recorded too loudly as in Figure 9-3. In this clip, you can see that the sound of the dog barking is peaking.

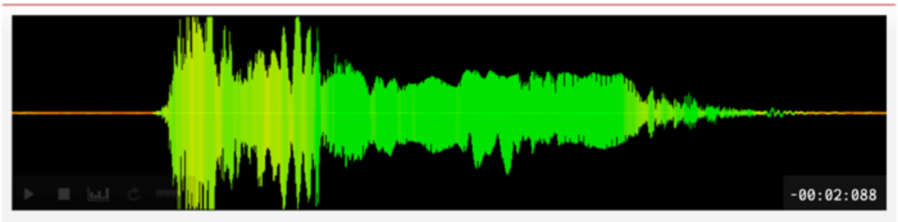

Figure 9-3. *An audio clip from freesound.org where the audio of the dog barking is peaking*

Figures 9-4 and 9-5 have been recorded at a healthy level.

Figure 9-4. *An audio clip from freesound.org that looks to have been recorded at a healthy level*

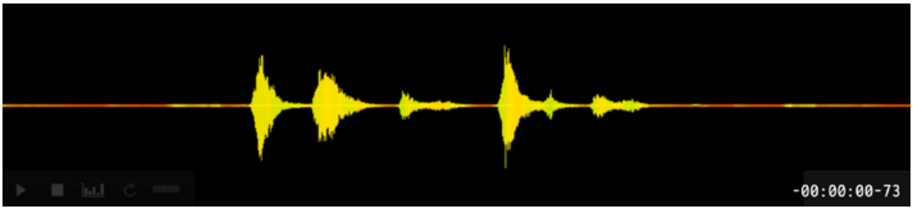

Figure 9-5. *A second audio clip from freesound.org that looks to have been recorded at a healthy level*

The visual representation of the audio file will give you an idea of whether a sound effect has been recorded well. The true test is listening to it.

Check the copyright information on sounds that you get off the Internet before using them.

Music

When we left our lovers, they were about to be launched into an interstellar adventure. But what kind of adventure? Action? Horror?

Using music, I have signaled that this story will be a schlocky drama that is guaranteed to have a happy ending.

Audio 9-3 Lovers in space – A touching drama

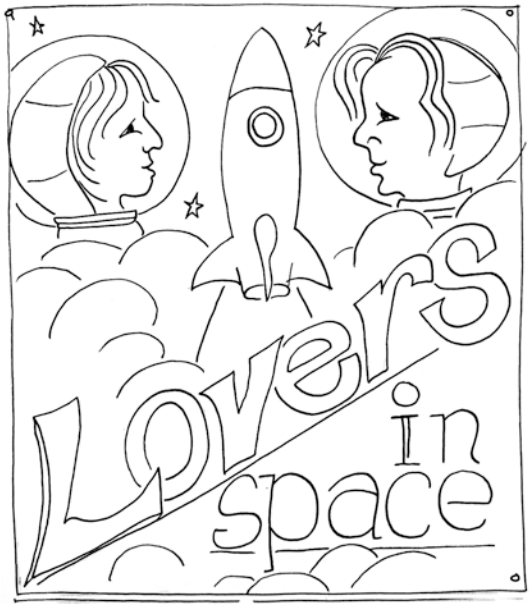

Figure 9-6. *Music, sound effects, and atmos situate my short radio drama in a spaceship, move the action along, and reassure the listener that there will be a happy ending*

Music is a time-honored way to heighten the emotional content of a story, but before you start adding it to your podcast, there are some copyright issues to consider.

Negotiating Music Copyright

There are a few ways to access music for your podcast without running into copyright issues. You can create the music yourself. If you know a musician, you can ask to use their music.

You can pay a copyright management organization for a podcasting music license. Copyright management organizations handle agreements covering the use of copyrighted music all over the world. For example, if you are making a podcast in Australia, you can pay an organization called APRA AMCOS to access music. They will distribute the royalties to the artists in Australia or elsewhere through their overseas affiliates. The **Online Mini Licence** should cover most Australian podcasts. Organizations that collect and distribute music royalties operate in most countries.

There are a number of different ways to access music outside of these copyright management organizations. You can find websites dedicated to music that is in the *public domain*. Music in the public domain is music that is so old that it's out of copyright. Some websites sell music that's *royalty free*. You will still need to pay for these tracks, and the organization that you pay will pass on the royalties. You might look at tracks with a *creative commons license*. These are works that musicians have released for people to use without payment in certain circumstances. These are usually only suitable for podcasts that are not intended to make a profit. You will have to read the fine print on the terms of use for any creative commons track. There are often conditions such as acknowledging the musicians in your work.

Now that you have found the music for your podcast, you will need to consider how it fits in with the other elements, especially speech.

Layering Music with Speech

When layering music under speech, you want to ensure that the speech is always discernible. Keep in mind that music will make it harder for someone to hear the speech if they're in a noisy environment or if they have compromised hearing. You can check speech intelligibility by listening to your mix quietly on your computer speakers. If you're listening back in a noisy environment, that is even better.

When I'm layering music and speech, I set the initial level on the music track and then manually turn it up and down using a volume envelope as in Figure 9-7. I err on the side of caution by setting the music to a level that is a little quieter than seems necessary.

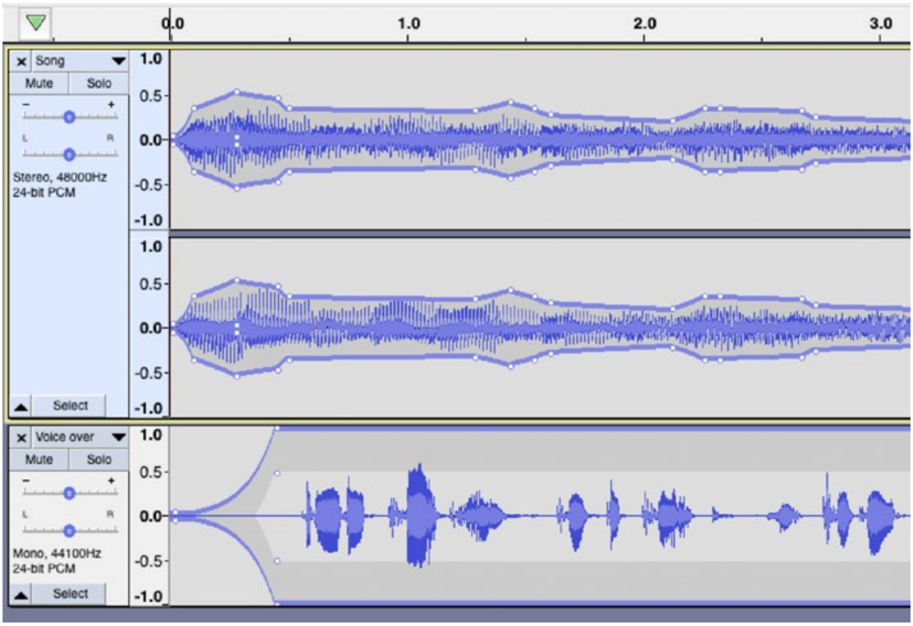

Figure 9-7. *Layering music and speech in Audacity*

In Figure 9-7, I have started the music at a medium loudness to draw attention to that element and then reduced the loudness for the duration of the dialogue.

John Phillips is a musician who is one half of Melbourne band *Not Drowning, Waving*, and has done a considerable amount of work creating sound tracks for film and TV. He says that there are certain types of music that work better layered under speech. A large part of speech intelligibility lies in the fairly quick, almost percussive sounds of consonants. Instruments that are percussive, such as drums, compete with these sounds. Instruments that make long and slow sounds like the bowed instruments don't compete as much with human speech. Similarly, music that's centered in the same frequency range as human speech will compete more than music that's not. The sound of a guitar is centered in the same frequency range as human speech, whereas the sound of a double bass is much lower. Following these two principles, the worst music to layer with speech is singing.

A short recurring motif can tie your content together or signal the start of a new topic. This can be the same short snatch of music, or it can be ongoing sections of one song. For example, you might layer the instrumental introduction of a song underneath the first section of voice-over. Then you could grab short sections such as a single verse and play each one between topics. Finally, you could layer the instrumental outro of the song under the final section of the voice-over.

Including music in your podcast can help maintain audience interest by breaking up long sections of dialogue, or it can be annoying. The audience might not like the music that you have included. Consider if the music matches the content. If, for example, your podcast is about music, then it would be great to play the entirety of a song that is being discussed. If your content is about accounting, then it might be more appropriate to play a short section of something fairly uncontroversial between topics.

Listening Back to Your Podcast

In the last chapter, I advised you to consider your audience when choosing the equipment with which you will play back your podcast during editing. During the editing process, you may have favored one type of equipment such as headphones for listening back. Toward the end of your editing process, it is useful to listen on different types of equipment to hear how successful your editing has been, especially if you are including layered elements such as music, atmos, and sound effects.

When listening back to your podcast, the two most important things to check are whether the speech is easy to understand and how well the different elements fit together. You can listen back to your podcast on headphones, out of the speakers of your computer, and even on decent speakers. Listen to your podcast loudly, softly, and at a medium level because, due to a quirk of human hearing, the volume will affect the prominence of different frequency bands. I find that the best way to know if I have successfully balanced different elements is to listen to the podcast quietly through my computer speakers. Then I will check that balance by changing the volume and listening through my headphones. If you listen to your podcast in a few different ways, you can feel confident that it sounds good on a wide variety of equipment at different volumes.

Listening through headphones, your computer speakers and even high-quality speakers are a good start and practical in terms of the editing process. If you really want to know how successful your editing has been, take your podcast out on a field test, the aim of which is to replicate your listeners' experience.

In the preface of this book, I shared an audio file, the second half of which was streamed through Apple Podcasts and played on a smartphone on a noisy balcony. I've repeated the file here as Audio 9-4. This is an example of a field test. Even though the quality of the audio in the second half of the file is significantly degraded, the speech is still easy to understand, so I consider my editing a success.

Audio 9-4 How your podcast might sound when played through the speakers of a smartphone in a noisy environment

If you take your podcast on a field test, try to imagine how your specific audience might be listening. Say, for example, that your podcast is aimed toward business people in Melbourne who you expect will listen during their morning commute. You can see from the analytics available on the site that you use to host your podcast that most of your listeners are streaming your podcast through Spotify. After you have uploaded your podcast, find a pair of low-quality earbuds and listen to it on a peak-hour train. Stream it through Spotify so that you can experience the changes that they've made to your audio. Listen to whether all of the audio can be heard without having to turn the volume up and down. Particularly listen to whether the speech is intelligible. You will quickly determine what elements of your editing have been successful and what elements need more work in the next episode.

Even if you don't take your podcast out into the field, it is important to listen to it through a streaming service at least once to hear how it sounds. If you don't know how to check the analytics on your podcast hosting website to see which streaming services your listeners are using, then choose Spotify. Spotify is becoming the most popular podcast streaming service in many major markets including the United States. Apple Podcasts is another good choice as they have dominated podcast streaming for many years.

If you know that many people are listening to your podcast through your website, there's a greater responsibility on your part to make the audio sound good. This is because your podcast won't undergo the compression processes of a major platform. How you upload the podcast to your site is how your listeners will hear it. It will be your responsibility to make sure that the quiet parts of your podcast will be loud enough to be heard in a noisy environment without the listeners' ears being blasted by the loud parts of your podcast. You will also want to check that the files that you have uploaded are small enough to stream effectively through your phone using phone data.

Figure 9-8. *Consider road testing your podcast. You could listen to it on a train through low-quality earbuds*

If you follow the editing processes covered in this chapter and the last, you can make podcasts that can be heard in a wide variety of environments through a wide variety of equipment. If you have made a clear recording, standardized the volume levels, and balanced the different elements well, then your podcast should be enjoyable to listen to in most circumstances.

Some Parting Words

Music, atmos, and sound effects are a great way to build an immersive world and connect with your audience. You can use these elements to affect mood and pacing and to increase your listeners' engagement with your content. Remember at all times to prioritize the intelligibility of speech to make your work accessible to the widest audience.

This is the end of the book. My hope is that in reading this book, you feel empowered to get out there and make a great podcast. Podcasting is a wonderful medium for communication. It's accessible, diverse, and it shows no sign of slowing down. I feel lucky that I've had so many interesting experiences through radio and podcasting, and I'm glad I've had this opportunity to share my passion with you.

You now have the opportunity to share your passion through podcasting. Optimizing your audio quality is giving your ideas their best platform. It's finding your unique voice and sharing it with the world. It's connecting with people like yourself or people who are completely different to you but want to hear what you have to say. It's raising awareness of an issue that's close to your heart. It's expressing that creative concept that could only come from you. Getting a handle on the technical side of podcasting means you can start creating.

Glossary

Acoustics

The qualities of a room or building that determine how sounds develop in that space.

Acoustic echo cancellation

In telephony, acoustic echo cancellation is a process which removes echo from a call.

Amplify

To increase the strength of an audio signal.

Amplitude

The strength of an audio signal.

Atmos

The ambient sound of any environment, such as a busy city street.

Audio

Generally speaking, audio is sound that has been translated into electricity. From there, it can be stored in various ways, manipulated, reproduced, and transmitted.

Bounce

When you are editing multiple audio files in a session in an audio editing program, you can *bounce* them into a single audio file. The single file will be referred to as the *mixdown*.

Bit rate

When converting analog audio to digital audio, the bit rate determines the detail with which the analog-to-digital converter stores each sample.

Channel

A channel is a pathway through which audio can travel. It's also known as a track.

C. M. Green, *The Podcaster's Audio Handbook*,
https://doi.org/10.1007/978-1-4842-7361-6

Compression/compressed

Compression is a process that makes an audio file smaller by removing data.

or

Compression reduces the difference between the loudest and softest parts of an audio clip. Audio that is compressed has had the peaks above a certain threshold reduced in amplitude.

Directional microphone

A microphone with a polar pattern that prioritizes sound from particular direction(s) and rejects it from other direction(s).

Dynamic range

The difference between the loudest and softest parts of a sound or a piece of audio.

Frequency

Frequency is like pitch. An airplane flying overhead makes a low-frequency sound and a whistle makes a high-frequency sound.

Gain

Gain is a bit like volume, except that it specifically refers to amplifying a signal.

High pass filter/HPF/low cut filter

A high pass filter filters out sounds below a particular frequency.

Input

An input is a pathway through which sound or audio can feed into audio equipment or computer software. It can be physical or virtual.

Latency/lag

Latency is the amount of delay between when audio enters and exits a system. A really high amount of latency is referred to as lag.

Level

A level is the strength of an audio signal. Ideally you should set your level at a high enough strength to record sufficient audio data, but not so high that it's peaking.

Mixdown

When you are editing multiple audio files in a session in an audio editing program, you can *bounce* them into a single audio file. The single file will be referred to as the *mixdown*.

Monitoring

Listening back to your audio.

Mono

A file format that contains only one channel, the center. See *stereo*.

Mp3

Mp3 is an audio file format. This file format is very compressed.

Omnidirectional microphone

A microphone that picks up sound consistently from every direction.

On-mic and off-mic

A person who is on-mic is speaking into the area around a microphone that is designed to pick up sound. A person who is off-mic is speaking into an area around a microphone that is designed to reject sound. See example (Audio G-1).

Audio G-1 On-mic and off-mic

Output

An output is a pathway for audio or sound to leave a piece of audio equipment or software. It can be physical or virtual.

Polar pattern

The polar pattern of a microphone determines the areas from which it picks up sound most effectively and the areas from which it rejects sound.

Processing

Processing is a general term that describes altering the characteristic of an audio signal. Examples of processing are compression and equalization.

Reverberation/reverb

Repeats of a sound that have bounced off surfaces near the source of a sound.

Routing

Routing is a term that describes moving audio from one place to another.

Sample rate

When converting analog audio to digital audio, the sample rate is the number of times per second that an analog-to-digital converter samples the analog audio waveform.

Signal-to-noise ratio

The *signal*, in the signal-to-noise ratio, is the amount of wanted audio signal. The *noise* in the signal-to-noise ratio is the noise that audio equipment makes as it operates. If you have a low signal-to-noise ratio, you'll hear a lot of noise and not enough signal. See audio example (Audio G-2).

Audio G-2 Recording demonstrating low signal-to-noise ratio made with a USB microphone

Sound check

A process of checking and calibrating audio equipment, performed before a recording.

Sound reinforcement system

A sound reinforcement system is an audio setup that involves microphones, speakers, etc., and is used to amplify sound and distribute it to an audience.

Stereo

A file format that has two channels, left and right. The two channels usually contain slightly different audio to give the impression of movement or of a physical space. See *mono*.

Track

A track is a pathway through which audio can travel.

or

A track is audio that has been imprinted on a storage medium.

Wav

Wav is an audio file format. This file format is uncompressed.

Voice-over

A voice-over is a recording of a person speaking that is intended to be explanatory or to tie different sections of audio together.

Index

Symbols

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