American Folklore Society

Field Recording of Interviews in Challenging Situations and Basic Tools of Audio Mixing

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Introduction

The purpose of this report to offer advice, guidance, and tips to American Folklore Society members who wish to learn about recording audio interviews in a variety of different environments, as well as those who wish to start doing some basic audio mixing of their own. It is primarily designed for those who wish to create professional-sounding productions for broadcast on the radio, via a podcast, on a soundtrack, or through some other media, but the guidance it contains should be widely applicable to anyone working in the realm of audio at a basic level.

This report builds on the earlier, excellent paper written by Taki Telonidis of the Western Folklife Center - "So What's the Story? A Primer on Making Radio". However, whereas that report covered reasons why folklorists might benefit from learning about radio production, gave advice regarding crafting stories, and shared guidelines as to how to set up a basic editing session, the nature of this report means that it focuses exclusively on technical matters relating to recording and basic mixing techniques.

To help illustrate some of the points made within this report, it contains a number of "before and after" audio examples. Please note that some of these require acute listening to discern the differences. Computer speakers are unlikely to be an adequate means of monitoring these changes; it is therefore either necessary to connect your computer to decent audio monitors or use headphones to listen to these examples.

Despite onslaughts over the year, from film and television and other media, pure audio remains a powerfully intimate and addiction means of conveying stories and talking about meaningful traditions

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This report is the product of the five days which I, Rachel Hopkin, spent training with Robin Wise, audio engineer, in Portland, Oregon. The guidance it offers is intended to be widely applicable. However, for reference purposes, here is a list of the key equipment used during training:

- Tascam HD-P2 Portable Stereo CF Recorder
- Shure VP88 Stereo Microphone
- 2 x Rode NTG-2 Shotgun Microphones
- Dead Cat Wind Muffs
- Avid Pro Tools 10 and Pro Tools 11 Audio Editing Software.

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RECORDING INTERVIEWS IN CHALLENGING SITUATIONS

What is a challenging situation when it comes to recording an interview? An aural purist might say it is any environment that is not a sound-proofed studio furnished with comprehensive, quality recording equipment operated by an ever-present, highly-trained audio engineer. However, even leaving aside the difficulties of finding such a studio, not to mention the costs incurred in booking it, most folklorists prefer to record interviews in an environment more conducive to an interviewee feeling at home and at ease. But what if that homely, easeful place were also a hub of extraneous and obtrusive background noise? Or what if some other element were present that could threaten the quality of the final recording, such as inclement weather if the interview were recorded outside?

Learning how best to deal with recording in challenging environments is important for all those working in the realm of audio. Attaining the best quality recordings in any given situation exponentially expands the manner in which your interviews can serve later on, including ways you may not have conceived of at time of recording. For example, you might embark on an interview expecting to use it only for your own ethnographic research, but then later decide to include it within a media production. That is only going to be possible if the recording is of decent quality. It is therefore essential to create the best raw material possible at the time of recording.

THE BASICS OF CREATING GOOD INTERVIEW RECORDINGS

In covering the basics of creating good interview recordings, it is necessary first and foremost to reiterate crucial advice from the Telonidis report: namely, that you need to get your microphone CLOSE to your interviewee (and to yourself too if you want your questions on tape). It is usually best to have the microphone within 8-10 inches of the speaker. This makes for a satisfyingly intimate sound and allows you to pick up on all the personality that their voice contains.

When positioning the microphone, avoid placing it directly in front of the speaker's mouth; instead locate it slightly below and to one side. Positioning it in this way has a two-fold benefit: it means that it will be out of direct sightline (and therefore less of a distraction) and also that popping "P"s and other harsh consonants are less likely to plague the recording.

Fixing the microphone to a boompole can help you achieve this proximity without encroaching on your interviewee's personal space, and it also means you run less risk of handling noise. A boompole is preferable to a microphone stand because stands are stationary while interviewees are not. With a boompole, you can subtly move the position of the microphone should the interviewee change stance.

Choice of Microphone

There is, alas, no such thing as a "one-size-fits-all" microphone. What you choose to use will depend on the equipment available to you, the circumstances of your interview, and the outcome you are hoping to achieve.

Using a **mono shotgun microphone** for an interview can be a good option. This type of microphone is designed to be highly directional. It focuses on the sound coming from the direction towards which it is pointed and therefore cuts out more ambient background noise that most other types of microphones. However, a shotgun microphone is also less forgiving if you hold it incorrectly in relation to your interviewee, or if s/he starts looking around the room while speaking, or if you are recording them on the move. In such cases, they are likely to go "off-mic" very quickly, resulting in a variable, more distant, or generally less-defined sound.

If you need your own voice (as the interviewer) to be recorded as well, then you will need to use two mono shotgun microphones and position them accordingly. In order to avoid microphone-handling noise, it is a good idea to attach them to a stereo bar and hold that with a hand grip, or to a boompole. Two well-positioned mono shotguns used in this way provide excellent separation of sound.

Another option is a **stereo microphone**. A stereo microphone will pick up on more background noise than a mono shotgun microphone so remember, the closer the microphone is to the source of the sound you wish to record – i.e. the speaker - the less distracting the background noise will be. If you are recording yourself as interviewer too, using a stereo microphone means that you only need one microphone. You need to position yourself close to your interviewee, and then judicious angling of the microphone back and forth can serve your recording well.

Most stereo microphones have an indicator telling you which side is picking up sound recorded on the left and which side right. Some models also have a range of settings that open or close out the ambit of the stereo sound picture which they record – for example, 60 degrees, 90 degrees, and 120 degrees. If you are recording just one voice, it is usually best to use the narrowest setting. If you are recording more than one voice, pick the one that best covers the placement of those being recorded.

Whichever microphone/s you end up using, avoid rushing into the recording. Spend time listening to how the voices sound and move your microphones around and adjust the recording level as necessary before beginning to ask your serious questions. This can serve as a good ice-breaking time and, in addition, the better you set up your recording gear at the outset, the better the final recording will be. Getting things set up right at the beginning takes only a few minutes, but attempting to diminish recording flaws caused by poor microphone placement or incorrect recording levels can take hours of studio time later on.

General Recording Tips:

- Always test your gear before you set off on a recording trip.
- Use “closed” headphones when recording. These allow you to monitor what is actually being recorded rather than what your ears are picking up from elsewhere. If you are not used to wearing “closed” headphones, they can take a while to get used to, but persevere; it is worth it.
- If you have a microphone that needs to be powered, either by battery or by phantom power from your machine, opt for phantom power when your recording device is connected to mains electricity. The phantom power will have a stronger signal and result in a stronger recording.
- Most microphones come with basic foam windshields. When placing this on the microphone, do not press it all the way down onto the microphone; instead leave a quarter to a half-inch of space. This can help prevent popping ‘p’s.
- It is a good idea to keep a couple of practical items in your gear bag: a set of pliers helps to tighten up microphone attachments and grips, and gaffer tape can be very useful to secure things in place.
- If you are travelling by plane, never pack microphones or recorders in checked baggage. Aside from the perils of delayed/lost suitcases, this kind of equipment is too delicate to reliably withstand the rigors of such transportation.
- Never leave batteries in your microphone or recorder when you are not using them. They are prone to leak and can irreparably damage your equipment.

CHALLENGING RECORDING SITUATIONS

Room Recordings with Ambient Noise

Ambient noise in a room can often be barely noticeable at the time of recording; later on though - perhaps when you are trying to knit the interview into a larger documentary - the sporadic hum from the radiator, fridge, air-conditioner, or the buzz from the lights or computer, or even the faint sounds of the water sprinkler in the garden next door, can become all too irritatingly obvious. The ideal is to eliminate as much of this noise upfront when you are actually making the recording. For example, I have, on many occasions, asked people to turn off their fridges or air conditioning for the duration of the interview. However, this is not always possible, convenient, or polite. So what to do?

- The first key tip remains the stalwart: record close. The nearer you are to your interviewee, the less noticeable the background noise will be.
- The second key tip is to position your speaker so that you are able to angle the microphone away from the source of the noise. So, for example, if the next-door-neighbor is mowing the lawn, do not have the speaker/s you are recording sitting right under a window that looks out onto said neighbor's garden.
- And the third key tip is record extra sound from the interview environment with no one talking. The resulting material is called "ambience", "room tone" or "wildtrack". When I trained at the BBC, I was instructed to always get two minutes of it. Two minutes is a very long time to sit there with no one talking, but I do still try to get at least one minute of it. You can use this ambience later in the editing process and by judicious fading, the transition from the interview in question to other tape will become a good deal less noticeable. When typing up your field notes or transcript, remember to make a note of where on the recording this ambience is.

(Incidentally, there is also some mixing know-how that can help in certain circumstances to diminish the impact of the unwanted sound. This is covered below).

Recording with Lots of Background Noise

Extraneous ambient room noise is one thing, but what if you are recording close to a building site or alongside a swimming pool? In certain circumstances, such background noise can add to the aural flavor of the interview and there can be advantages to having an interesting background noise, especially if it is relevant to the subject. But usually this is a less-than-ideal situation.

- As before, record close to your interviewee.
- Record at least several minutes of ambience so that later during the mixing process, you can fade in and out of that particular aural environment without the sound change being too abrupt.
- In terms of microphone choice, mono shotguns can be a good option as they cut down on the background noise. However, if you think you might want the background noise to provide aural interest, consider using a stereo microphone. As you record, make sure that your interviewee is in the center of the aural picture. If your stereo microphone is one that has an adjustable range of audio pickup (see above), do not make its recording range so wide that there is a recording hole in the middle. If using a stereo microphone in this situation, I would opt for recording the interview itself using the narrowest range but also record some extra background noise, when no one is speaking, using a wider range. Then I would knit the two together during the editing/mixing process in order to make a richer sound picture.

Recording in Windy Situations

Recording outside when it is windy is challenging. Whilst the sound of wind can add to sense of place, it can also become intrusive and distracting. Most microphones come with a foam windscreen but these are not usually able to withstand anything more turbulent than a mild breeze. Here are some things you can do if you have no choice but to record in the wind:

- Move to the side of building, behind a bush or a tree, or anywhere that helps to block the wind.

- Crouch down with your interviewee as you talk, with both of you huddling around the microphone in such as way that your bodies provide a kind of windbreak.

- Buy a windjammer. A windjammer is specially designed to combat wind-on-microphone noise. Often known as dead cats, these fake fur covers can be relatively effective, especially when you use both the windjammer and the foam windscreen together. Incidentally, avoid the temptation to smooth down the fur on the windjammer; the fluffier and more upstanding you can make the fibers the better, as the wind impact is diminished by the strands of fur.

- If you are going to be doing a lot of recording in potentially windy situations, consider investing in a microphone blimp/zeppelin windscreen. They are pricey but are the most effective tool for combatting wind-noise. If you are a practical kind of a person, you might be able to fabricate a version for yourself. This webpage and this one offer some tips.

(Note: some sound engineers favor omnidirectional microphones in windy situations as they are reputed to be less susceptible to wind noise. However, Robin Wise does not use omnidirectional microphones precisely because of their lack of specific directionality, and therefore they are not considered within this report.)

Recording in the Rain

A chief concern when recording in the rain is potential damage to your equipment. You might be able to safely tuck your recorder out of harm's way in your bag, but your microphone is going have to be out and potentially exposed to harmful elements.

- To protect the microphone, place a non-lubricated condom over it and secure it with a rubber band, then place the windscreen over that. Make sure you later allow your damp windscreen to air dry; if you leave it damp in your bag, it may go moldy.

- In terms of the recording itself, the rain is going to create a background noise. That is not necessarily a bad thing, but you or your interviewee should explain on tape what is making the noise as heavy rain, in particular, can sound like a surprising number of other things, including the sea waves and applause.

- As always, record close and record ambience.

Recording Multiple Speakers at the Same Time

I recently went to the house of an elderly lady intending to interview her and perhaps her son as well. When I arrived, I was greeted by a room full of the lady's relatives all wanting to talk about the family's dance tradition. I do not like doing collective interviews and the most people I had ever previously interviewed together was four. Here, I had close to twenty scattered over three sofas and numerous chairs. What to do?

- Explain to your interviewees the limitations of your equipment: namely, that you will record them best when the microphone is close to them. So please, when they want to speak, indicate their willingness (perhaps by waving) then you can move swiftly towards them.

- Locate yourself as much as possible in the center of the group and keep on your feet so you can move around.

- Use a stereo microphone if you have one.

- Fixing your microphone to boompole will help you quickly gain proximity to your speakers.

- Begin to ask your questions. With each one, wait to see who seems ready to respond, then move the microphone close to them. If more than one person looks ready to speak, make a mental note and go to them one by one.

- If a speaker begins to talk before you are able to get close to them, use judgment to determine whether to ask them to start over by taking into consideration how far “off mic” they are, how deeply they are into the flow of whatever they are talking about, and so on.

In the particular case I outlined above, I followed these steps and then, once all the group questions were complete, I politely asked the rest of the relatives to allow me to interview the lady alone. Some asked remain in order to listen in, which was fine, and it was much easier to focus on my questions to the lady at that point.

Basic Tools of Audio Mixing

Mixing, for an audio producer, refers to the production phase during which various aspects of the recorded sound are manipulated, such as the volume levels, frequency content, dynamics, and panoramic positioning of the sound, along with the addition of effects such as reverb, to produce an aurally seamless final piece. It is usually something one undertakes when creating some kind of public presentation such as a radio program, a podcast, or a multimedia soundtrack. That said, there could be circumstances in which you might choose to apply certain mixing tools to individual recordings simply in order improve their sound. Or, conversely, you could use them to improve the quality of sub-optimal recordings that you have not recorded yourself, but perhaps taken from an archive or other audio repository.

Mixing can be a very creative process, particularly as you become more experienced. It is quite easy to find yourself spending hours deciding how and when to fade in and out of a particular piece of music, or how to use which sound effect. But it also has more practical and prosaic ramifications by helping to ensure that a listener's attention is not drawn to the fact that one of your interviewees was recorded at a high level and another at a low level, or that there was a lawnmower in the background on one recording, or an airplane coming to land at a certain point during another. The focus of this report is to provide guidance on some tools that help with these basic aspects, rather than offer tips on the more creative aspects of mixing.

If you can mix audio pieces to a decent level and thereby deliver a professional sounding final piece, the range of possibilities for how that piece might be used grow. A local radio station with limited resources is far more likely to take a feature from you that is "good to go" rather than if you deliver something they need to spend several hours bringing up to broadcast level. It also means you can post your audio on other platforms - such as websites - without having to contract out to a dedicated audio engineer.

Mixing involves technical know-how, but also awareness, judgment, and experience. As a result, mixing and ear training go hand in hand. It takes time to learn how to judiciously and tastefully fade in and out of music and/or to feel when something is too loud or too soft. (By the way, in audio engineer-speak, loud = "hot".)

Any audio editing software worth its salt will have some mixing capabilities. You may have to experiment a bit with your particular software to work out how to use it, but then the guidance offered below should provide some basic guidance no matter which package you are using.

Basics of Mixing

- When you mix, organize your workstation so that you can listen back to your work via a pair of decent monitors, which excludes computer speakers. Mixing using headphones is not ideal UNLESS the piece you are mixing is only going to be heard via headphones (such as an audio tour designed to be heard via a smartphone).

- It is best practice to fade in and out of every single clip you are using in your mix. You do not want the listener's attention to be caught by changes in room sound or background noise, even when the change might seem almost imperceptible.

- A good rule of thumb is to imagine that your target audience is listening to your piece via high-end monitors and tailor your work accordingly.

- When you think you have a mix pretty close to finished, take a break from it, then come and listen back to it in a different way. (I usually do some ironing at this stage.) Changing the way in which you listen can help you gain a different perspective on how your work is really knitting together. Make a note of everything that sounds jarring and then fix it later.

Sound Levels
How loud ("hot") or soft is a piece of audio? How loud or soft do you want it to be? This section addresses the levels of sound volume.

- When you start altering the volume of your recording clips, you need to make sure you are adjusting them to play back at acceptable broadcast level. Acceptable broadcast level refers to the decibel (dB) measurement of your audio and it can differ from country to country and from broadcast network to broadcast network.[1] Robin Wise suggests that the average level of audio – unless advised otherwise – should be between -9 dB and -12 dB and peak at -3 dB. Nothing should go over 0 dB as that is the point at which sound distorts.

- In order to find out the level of your audio, you need a meter reader. Some audio software programs have one built in but if yours does not, Sonalkis offers a free meter plug-in that works with a variety of software packages and you can find it here: http://www.sonalksis.com/freeg.htm.[2] Use it according to its directions to check for average and peak meter readings.

- When you introduce your first piece of audio into the mix, set the volume of it to broadcast level, then adjust the playback volume dial on your monitors to the point where you can comfortably listen to it and **maintain the dial at that setting**. In this way, you will train your ear to recognize how things sound at broadcast level. After a while, you will probably find that you do not need to use the meter reader to check every single recording you bring into the mix.

- Assuming you have various different elements of audio in your mix, the next step is to work on balancing your different elements of sound so that none of them sounds too loud or too soft compared to the others. At this stage, it is actually not a good idea to rely too heavily on the meter level, as the way we hear things does not always correspond to actual dB levels. For example, women's voices use higher frequencies than those of men, and higher frequencies are easier to hear, so even if the meter is telling you a woman is speaking at the same level as a man, you may nonetheless want to bring her level down to compensate for the higher and easier-to-hear frequencies in her voice. (This is what I mean when I write that good mixing is about awareness, judgment, and experience!)

- Once you have all the different elements of sound in your mix balanced in relation to one another, it is a good idea to install a Master Fader. A Master Fader is available as part of most audio editing software packages and it monitors the collective output from all of the tracks in your editing session. So, for example, if you have someone talking over the top of music, the speaker alone may be at broadcast level, but the addition of the music may make the overall sound too loud. The Master Fader will give you the sum of all tracks mixed together and if necessary, allow you to adjust the overall sound level in order to maintain acceptable broadcast level.

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[1] A decibel (dB) is the unit measure of intensity of sound
[2] In a software package, a plug-in adds a specific capability or feature to an existing application.

Equalization of Frequencies
Equalization, or “EQ-ing” is one of the most commonly used mixing tools. It refers to adjusting the balance of a signal by “boosting” (increasing) or “cutting” (decreasing) certain frequencies. Frequencies are measured in hertz (Hz) and the higher pitched a sound is, the greater its Hz measurement; the lower it is, the lower its Hz measurement. Employing EQ means you can enhance frequencies you want to be more prominent and reduce those that contain unwanted noise. It is a means whereby you can try to diminish the impact of extraneous or unwanted noises on your recordings, such as car engine noise, gratingly harsh consonants, electrical hum, and so on. Most audio editing software packages come with some EQ capability built in. Please consult your software guide to find out how to access this capability.

Here are some important points to remember when it comes to EQ:

- Identifying frequencies for EQ purposes takes trial, error, and practice. Some audio software packages contain frequency analyzers that can help pinpoint the frequency range of a given noise. If yours does not, you can purchase one separately, or try Bluecat’s free analyzer here: http://www.bluecataudio.com/Products/Bundle_FreewarePack/ and follow instructions. It is basic but better than nothing.

- If an unwanted sound is affecting an entire sound file, apply the EQ to the file as a whole. For example, if there is an air conditioner hum in the background of an entire interview, apply appropriate EQ to the entire file.

- If an unwanted sound is only present during a portion of the recording, or if the recording blighted only, for example, by a few popping “P” consonants, then isolate the offending section/s and apply the EQ only to them

- Whenever applying EQ, use caution. Too much EQ can quickly adversely affect the voices recorded and make your audio sound unnatural.

- At times, there may be several unwanted noise issues going on at once; in that case you may need to use several different types of EQ.

- The key with EQ is practice. Experiment with different gradations of frequencies and see how the result sounds. Compare the original sound with the EQ’d sound. When you have found an EQ’d version that you think seems close to the original but in which the bothersome noise is less attention-grabbing, take a break and do something else, then come back and have a fresh listen. Are you still happy with it? If so, keep it, otherwise continue experimenting.

EQ and Common Offending Sounds

Reducing Low Rumbling Noises
Low rumbling noises include car engine hum, reverberation caused by a speaker's voice bouncing off glass and plastic in the recording environment, microphone handling noise, wind noise, and background traffic. One possible tool to apply in these cases to diminish the offensive sound is a **"high pass filter"**. A high pass filter allows the high frequencies to pass through unaffected, and only impact the low frequencies. Another option is a **"peak filter"** which allows you to increase or attenuate a particular frequency range.

Although there is no firm rule, a high pass filter is generally used on frequencies of 120 Hz and below. Here are some typical problem noises and the frequencies that you might look to reduce:

Car engine hum	c. 100-120 Hz
Wind noise	c. 80-100 Hz
Room reverberance	c. 100 Hz

Remember to experiment though. For example, for the car engine hum, you might try reducing the frequency at 90, 100, 110, and 120 Hz to see which produces the better result.

Here is an example of a recording in which the car engine noise is prominent behind a speaker.

INSERT "AFS – Daniel in Car – Before"

And here is the clip again with a high pass filter applied at 106 Hz

INSERT "AFS – Daniel in Car – After"

Here is another example, this time of a man's voice reverberating within a small room which has created a booming sound

INSERT "AFS – Milan Interview – Before"

And here is the clip again, with a Peak Filter applied at 100 Hz. The result is a reduction in the boomy sound.

INSERT "AFS – Milan Interview – After"

Reducing Hiss Sound
Hiss sounds on recordings are often caused by sub-optimal recording gear (although there can be other factors). Hiss is often a broadband sound, meaning it permeates across the entire frequency spectrum, although it often seems to be loudest in the high frequencies. As a result, a **"low pass filter"** can be a good tool as it allows the low frequencies to remain untouched, whilst diminishing high frequencies. An alternative is again to use a "peak filter".

High hissy sounds usually fall in the 7000-10,000 Hz range.

In this example, a strong background hiss is present which was probably caused by a microphone that did not put out a good signal with a recorder, or possibly by a bad cable.

INSERT – "AFS – Hiss - Before"

Here is the clip again, after EQ has been applied, using a Peak filter at 10,000 Hz.

INSERT – "AFS – Hiss - After"

The result is a diminution of the hiss though it still remains very present because it was so loud on the original recording.

Reducing Electrical Hum

Electrical hum is generated by a recording device picking up on the sound made by other electrical equipment, such as lighting. Because this kind of sound comes in only a very narrow band of frequency, the tool needed to reduce it is called a **"notch filter".** A notch filter is designed to sharply attenuate only one specific frequency and leave the surrounding frequencies unaffected.

In the US, electricity runs on 60 vibrations per second and the frequency of the sound it causes may be at that frequency, and also at the frequency of the overtones above it (which effectively means they impact the frequency figure times 2, times 4, times 8 etc.)   You may therefore need to use the notch filter several times over, targeting 60, 120, 240 and even 480 Hz. (Note: if you are recording in a different country, the electricity may run at different frequency.)

In this example, I recorded myself standing next to a refrigerator:

INSERT "AFS – Rachel and Fridge – Before"

Here is the same clip with a notch filter applied targeting 120 Hz.

INSERT "AFS – Rachel and Fridge – After"

The hum is still there but it is subtly diminished. To reduce the hum further would have resulted in a notably negative impact on the voice, hence this clip serves as a reminder as to why it is better not to record next to such an apparatus in the first place.

Using EQ on Voices:-

The intention when using EQ on voices is not to change the speaker's true sound, but to correct any distortions caused by a negative interaction between the voice and the microphone/recorder/recording environment.

Reducing Sibilance

Some recording situations result in an exaggeration of a speaker's "S"s on the final tape – either as a direct consonant (in words like "Susan", "sausage", "sassafras") or as part of a "Sh" sound combination ("sugar", "shaman" etc). EQ to can diminish this sibilant impact In fact, "S"s so often cause offence that there is a mixing tool designed especially to help diminish them called a **"De-Esser"**. Be careful with it though. If you are too aggressive with the De-Esser, you may leave your speakers with a lisp!

The frequencies to reduce tend to be as follows:

Male “S”	c. 5500 Hz
Male “Sh”	c. 3375 Hz
Female “S”	c. 6800 Hz
Female “Sh”	c. 5000 Hz

Here is a clip of me in which I really exaggerate the “S” sounds:

INSERT “AFS – Sibilant Rachel - Before”

Here is the same clip after the De-Esser has been applied targeting 6800 Hz.

INSERT “AFS – Sibilant Rachel - After”

Because I had so exaggerated my S sounds, the “after” version is still notably sibilant, but the impact is slightly diminished.

Reducing Muddiness

Sometimes a speaker’s voice records in such a way that it ends up sounding deeper and less clear on the recording than it did in person. To brighten up such recordings, use a “high pass filter” and start to look to reduce frequencies in the following ranges:

“Muddy” male voice	80-120 Hz
“Muddy” female voice	90-130 Hz

Here is an example of this phenomenon:

INSERT “AFS – Muddy Gibford - Before”

In order to make speaker’s recorded voice sound more like he did in reality, I used a high pass filter at 120 Hz with this result:

INSERT “AFS – Muddy Gibford - After”

Reducing Popping “P”s and Cracking “K”s

Popping “P”s and cracking “K”s occur when the speaker aspirates into the microphone in a manner that has a similar effect to wind noise. To help combat this sound, you need to work only on a very small segment of audio. Isolate the offending portion then use a “high pass filter” to rewrite just that small clip of audio. The frequency range you should reduce will be in the following ranges:

Popping “P”s	200-250 Hz
Cracking “K”s	250-350 Hz.

Another trick to help reduce the sound of Ks in particular is to isolate the K sound and do a slight fade in at the beginning of its impact.

In this short clip, the speaker's "P" pops on the "posits" of "deposits", has a slight cracking "K" on the word "county", and a worse "K" on the word "course".

INSERT "AFS – Carolyn P and K – Before"

Addressing these harsh consonants involved a combination of EQ and fades. The "P" was treated with a high pass filter targeting 220 Hz and the "K"s with a high pass filter aimed at 250 Hz as well as a couple of small fades. Here is the result:

INSERT "AFS – Carolyn P and K – After"

Other Mixing Matters

Panning

In stereo, panning refers to the spread of sound in the left channel and the right channel. With panning you can control where sound happens within the two channels and judiciously panning helps to create an interesting stereo picture.

For example, imagine that you recorded interviews with two different speakers both in mono, then you edit them together to that you have one speaker talking first, then the second, then back to the first and so on. Rather than having both voices come back at you from the exact same central point in space, you might choose to have one speaker slightly off to the left and the other slightly off to the right. This makes sense if you think about it, because in a normal conversation you never hear people speaking to you from exactly the same point. With two voices, a good rule of thumb is to have them situated at 10 to 2, as on a clock-face, where the center is 12. Consult your software guide to find out how to pan.

Here is the start of a piece I made about the Cowboy Poetry Gathering in Elko in 2014 prior to applying any panning:

INSERT "AFS – Vox - Unpanned"

And here it is again, this time with panning applied - you will need to be listening in stereo to appreciate the difference. I have left the first voice you hear, as well as my own, centrally panned, and then the others have each been panned either off to the left or the right.

INSERT "AFS – Vox - Panned"

If you were applying panning for effect in a longer piece in which you repeatedly returned to the same speakers at different points in time, make sure you keep them panned in the same place. In this example, the third woman who speaks returns after the man's voice, and in each case, I have placed her slightly to the left. It is also convention to have the main speaker or presenter centrally panned which is why my own voice on this clip – which is the start of a much longer report – is in that position.

Putting It All Together

When it comes to putting together all the different elements, it really comes down to judicious balancing of the sound to make sure all the pieces mesh together, with nothing sounding too loud or too soft, too abrupt or too drawn out. This takes a lot of experience, expertise, and practice but the processes outlined above serve as basic tools.

Here is an example of how the start of a piece I made for the BBC sounded prior to being dealt with in any way. There are no fades or crossfades or adjustment of levels, let alone EQ-ing or anything of that ilk. It was called *Musical Migrants: Nashville*. Unmixed, it is virtually impossible to listen to:

INSERT "AFS – Musical Migrants Nashville - Unmixed"

Here it is after the mix, which involved various types of EQ, a lot of playing around with levels, a lot of fades, and crossfades to cover up some tricky speech and music edits:

INSERT "AFS – Musical Migrants Nashville - Mixed"

So as you can hear, a good mix makes a HUGE difference.

THE EQ SETTINGS CRIB SHEET

EQing is an art form and nothing is black and white. It is necessary to use and experience the filters to fully understand them. However, this sheet provides some pointers about the filters you might try using and the frequency ranges that you might seek to impact with them.

Reducing Low Rumbling Noises:

Try using: High Pass Filter or Peak Filter
Possible frequencies to target**:**

Car engine hum	c. 100-120 Hz
Wind noise	c. 80-100 Hz
Room reverberance	c. 100 Hz

Reducing High Hissy Noises:

Try using: Low Pass Filter or Peak Filter
Possible frequencies to target**:**

7000-10000 Hz range.

Reducing Electrical Hum:

Try using: Notch filter
Frequencies to target for recordings made in the USA:

60 Hz, 120 Hz, 240 Hz and 480 Hz.

Reducing Sibilance:

Try using: De-Esser
Possible frequencies to target**:**

Male “S”	c. 5500 Hz
Male “Sh”	c. 3375 Hz
Female “S”	c. 6800 Hz
Female “Sh”	c. 5000 Hz

Reducing Vocal Muddiness:

Try using: High Pass Filter
Possible frequencies to target**:**

“Muddy” male voice	80-120 Hz
“Muddy” female voice	90-130 Hz

Reducing Popping “P”s and Cracking “K”s

Try using: High pass filter only on offending portion of audio, not across the whole recording.
Possible frequencies to target**:**

Popping “P”s	200-250 Hz
Cracking “K”s	300-350 Hz.

CONCLUSION

Recording in challenging situations, mixing and EQ - all these things take a lot practice. When I worked at the BBC, we producers were blessed to work with dedicated in-house sound engineers who were rigorously selected and had gone through months of dedicated training. However, few folklorists have access to such personnel luxuries and I hope therefore that this paper has given you some tips on how to get started on your own.

Although I have covered some tools which can help improve poorly recorded tape, it is worth stressing that the better you can make your recording quality from the outset, the better. The tools outlined above may help to improve the quality of poor tape but only to a limited extent. In addition, applying EQ can be extremely labor intensive, whereas you might only have needed to spend a few minutes at the start of your original recording to make sure you were make you had adequately set up the recording situation.

It is also worth suggesting that if you have a complex mix to undertake – one that includes multiples voices, music, effects or other sounds – until you have had the opportunity to practice extensively all the elements of mixing, do not discount the idea of building some professional audio engineer hours into your budget, should your circumstances permit. If a top notch audio production is your goal, then a professional can be worth his or her weight in Tascams, Sonys, Zooms, and any other recording devices you care to name. Not only do such personnel have expertise and experience, they are also likely to have invested many thousands of dollars on a range of sophisticated software that allows for sound manipulation and improvement way beyond the basic plug-ins that come as standard with most audio editing software.

Despite many onslaughts from film and television, etc., pure audio remains a powerfully intimate means of conveying information and an excellent medium for folklorists. I do hope you have found this paper useful.

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