High Quality Podcast Recording

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Abstract

A good podcast comes with great content as well as great audio quality. Audio quality is as important for podcast as is layout, style, and formatting for documents. Just as these parameters influence readability, audio quality influences listenability. If you want to get a decent-sized audience as well as (potentially) interesting guests you have to make sure your content is presented adequately. Audio quality is an important ingredient here. This document describes the audio equipment used in the following podcasts:

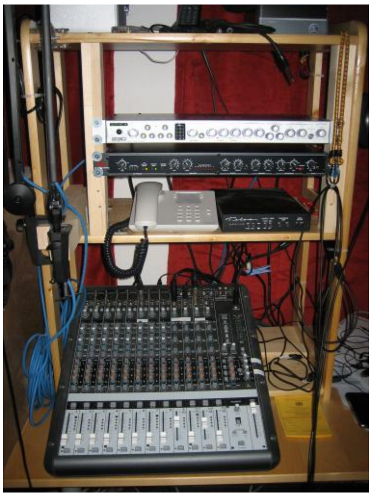
- omega tau - science and technology in your headphones
  http://omegataupodcast.net
- Software Engineering Radio - the podcast for the professional developer
  http://se-radio.net

Note that today most well-known podcasts have really high production value, including professional audio quality. Some of the podcasts describe the equipment and the recording process, most don't. This document describes my approach to recording a decent quality audio. While it is not as professional as what Leo Laporte does [1], is also much more affordable. I hope this helps new podcasters get up to speed faster than I ☺

Environment

The most important ingredient to good audio quality is the quality of the raw audio. An important parameter here is the environment in which you record. Obviously, it should be as quiet as possible. That also means that you should make sure your computer does not produce excessive ventilation noise. There are silencer kits, although I don't use one of these.

The next important consideration is the dryness of sound. The sound is dry if it has no echo or other effects applied to it. For the recording this means you should try to avoid any echo whatsoever. This is surprisingly hard to achieve outside professional studios. An office room with lots and lots of books and carpet and stuff is a good start. The inside of cars is also good! You can also use specific dampers to achieve this effect. For example, I use a couple of HOFA audio absorbers [2] to dry up my voice.

More recently, I am also playing with putting an old mattress behind me when recording.

Recording Equipment

Microphone

The next step for good audio is the microphone and recording equipment. As for the microphone I use a Heil PR 40 [3] together with a pop filter from BSW [4]. I also use a suitable microphone arm [5] and a spider web microphone mount from Heil. This makes sure that the microphone is free from movement noise from the surrounding.

Voice Strip

We use two different microphone processors (aka voice strips) for our two microphones. One is the dbx 286a [6].

This device has a couple of features that improve the sound of voice. The following table briefly explains these features and the settings I use.

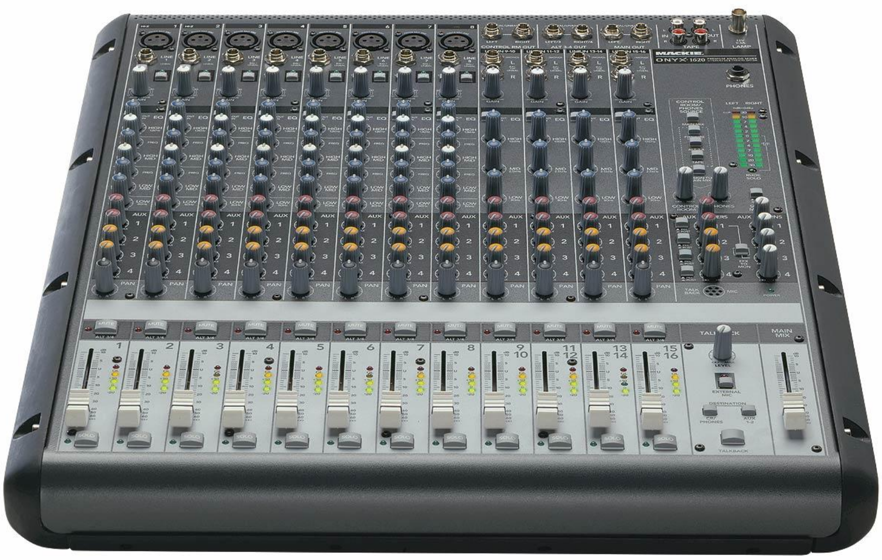
High-Pass	Removes sounds below 80Hz	On
Compressor	Used to adjust the volume of your voice: increase quiet stuff,	Drive: 5 Density: 4

	dampen loud parts.	
De-Esser	Removes the loud hissing sounds produced for example by s-sounds.	Frequency: 5k Threshold: 6
Enhancer	Basically an equalizer	LF Detail: 5 HF Detail: 3.5
Expander/Gate	Removes very very quiet sounds and zeroes it.	Threshold: -15 Ratio: 1.5:1
Output	Output Gain (make sure it never clips!)	Gain: +8

The other one is an Aphex 230 [15]. It basically does the same thing as the dbx, but has somewhat better quality and some additional options.

Mixer & Recording

As a mixer, I used the Onyx 1620 [7]. In retrospect, I could have used a smaller one, but to be able to record two voice channels, Skype, and/or telephone I recommend at least 4 to 8 channels. Using the various Aux-out options, I can create several different mix-minus configurations for the various incoming lines.



I use the Onyx FireWire plug-in [8] to feed the audio from the mixer into the PC on which I record. Note that this device feeds every channel separately into the PC. It is absolutely crucial that you record every channel separately to be able to adapt various audio parameters in postproduction for each channel.

The actual recording is done on a PC running Adobe Audition [13] based on the Onyx FireWire driver. Recording could also be done with the free Audacity tool. However, for editing, Audition is significantly better.

Cables

From time to time I hear that good audio cables are important for good sound. I have to say I cannot confirm this. While I don't use the cheapest cables, I don't use anything fancy either. You might want to make sure you don't have any unnecessary adapters or plugs, though: in my previous mobile setup I had used XLR to 1/4'' stereo jacks and this actually did produce noticable noise.

Recording from Skype

If you want to record from Skype, there are various Skype recorder software packages. However, my experience is that external recording equipment is better. In my case I simply route the line-out of the notebook that runs Skype into one of the line-in channels of the Onyx mixer.

Skype can produce very good audio quality if you're lucky and if you stick to certain rules. All of them are explained perfectly in this video [9]. Apart from the obvious stuff such as shutting down everything else on your machine, especially the thing about a dedicated fixed port (which avoids intermediaries in the packet transport) seems to be very important.

Also, you want to make sure that your guest at the other end uses an acceptable microphone. 20 EUR USB headset microphones are good enough, the built in microphones of most notebooks aren't.

Recording from the Telephone

In many cases you will also want to record the good old telephone. To do so, I use a Telos ONE digital telephone hybrid [10].

It captures "the other end" of the telephone conversation and feeds it into the mixer. It also feeds my own voice signal from the mixer into the telephone. The setup produces really good quality if the telephones you use are connected via real land lines. Voice over IP phones (found more and more often in corporate environments) will result in significantly reduced quality.

Mobile Equipment

To record conversations outside of my studio I use a Zoom H-4n [11] as the recording device. It can record four tracks over all simultaneously: two external microphones, and two built-in microphones. It creates uncompressed WAV files. It also has a built in compressor.

I use two AKG C 520 [12] headset microphones will pop filters; they can be plugged into the H-4n via the XLR connectors directly.

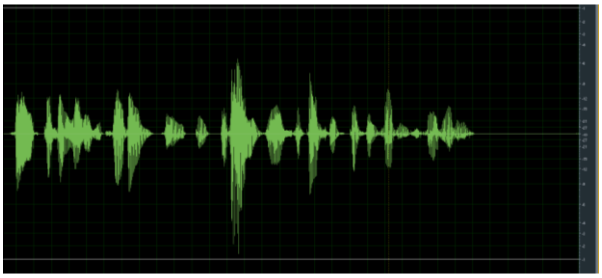
Using headset microphones is important to make sure the speakers don't move the microphone away from their mouth while talking. Make sure the microphone does not touch the speaker, and put it outside the mouth/nose breathing airstream.

Postprocessing

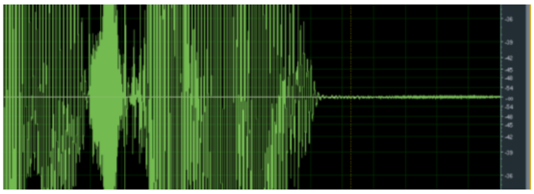
If you record the raw data in the way explained above you will not have to do a lot of postprocessing (of course you have to do content editing, but not a lot of sound stuff).

The only thing I do is adjusting the levels to make sure that the voice has an even level and is pleasant to listen to, especially in noisy environments where podcasts are often consumed. Let's look at an example.

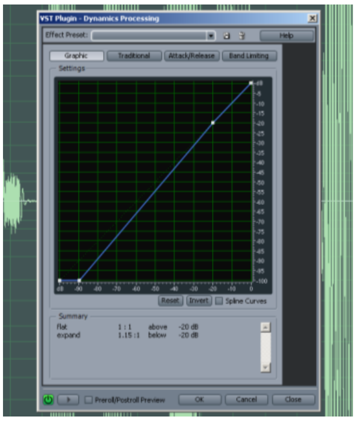
Here is the waveform of a raw recording. Notice the different levels (amplitudes). We want to make sure they are more even after postprocessing.



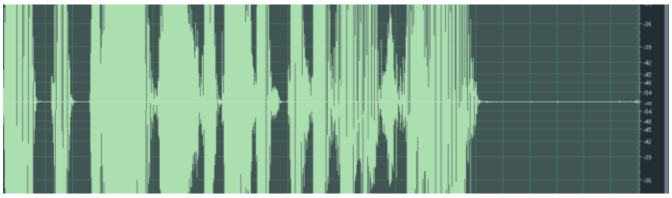
If you zoom in you can see that the parts that look quiet really are not completely quiet (which will be a problem when we use the limiter later).



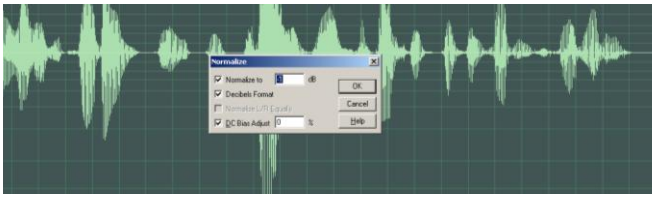
So the first thing I do is to run the dynamic processor to remove everything below -90 dB (sometimes also below -80 dB if there is more noise). Try to avoid using an actual noise reducer since it usually produces artifacts.



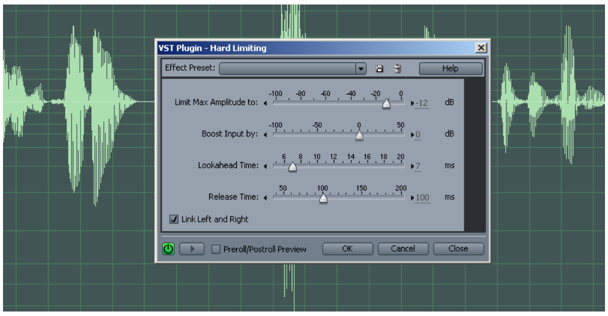
As you can see, this makes quiet parts really quiet.



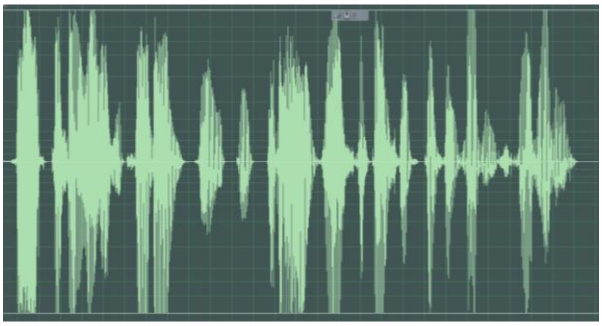
I then use the normalizer to "pull up" the max amplitude to -1 dB, exploiting fully the available dynamic range.



I then use the limiter to cut away everything above a certain dB limit. Depending on the raw material this can be anywhere between -3 and -15 dB.



I then normalize again to -1 dB. This results in a relatively homogeneous amplitude. In the example at the right you might want to do the limit/ normalize sequence once again, for example at -4 dB.



Mixdown

There's not much to consider during mixdown. If you use the postprocessing approach shown above for all the constituent parts of the final mix, the overall volume should be fine (if not, you can always run Levelator [14]). Here are a couple of things to consider:

- Sometimes you want to adjust the overall level between the tracks. For example, things recorded via the phone aren't perceived as loud as locally recorded material (phone recordings have less base!). So you might want to boost the phone track by 2 or 3 dB (or lower the local track by that amount).
- If you use music, especially highly compressed rock music, make sure you reduce it by 3 to 4 dB since it is always perceived to be louder than the actual numerical dB value suggests.
- Finally, if you have a dialog of two people, you might want to use -15/+15 stereo panning.

References

[1] http://wiki.twit.tv/wiki/Equipment

[2] http://hofa-akustik.de/pages/produkte/absorber.php

[3] http://www.heilsound.com/pro/products/pr40/index.htm

[4] http://www.bswusa.com/proditem.asp?item=RE27POP

[5] http://www.heilsound.com/pro/products/pl2t/

[6] http://www.dbxpro.com/286A/286A.php

[7] http://www.mackie.com/de/products/onyx1620/

[8] http://www.mackie.com/products/onyxfirewire/

[9] http://www.blogarithms.com/index.php/archives/2007/12/23/skype-for-interviews/

[10] http://www.telos-systems.com/one/default.htm

[11] http://www.zoom.co.jp/english/products/h4n/

[12] http://www.akg.com/site/products/powerslave,id,984,pid,984,nodeid,2,_language,EN.html

[13] http://www.adobe.com/products/audition/

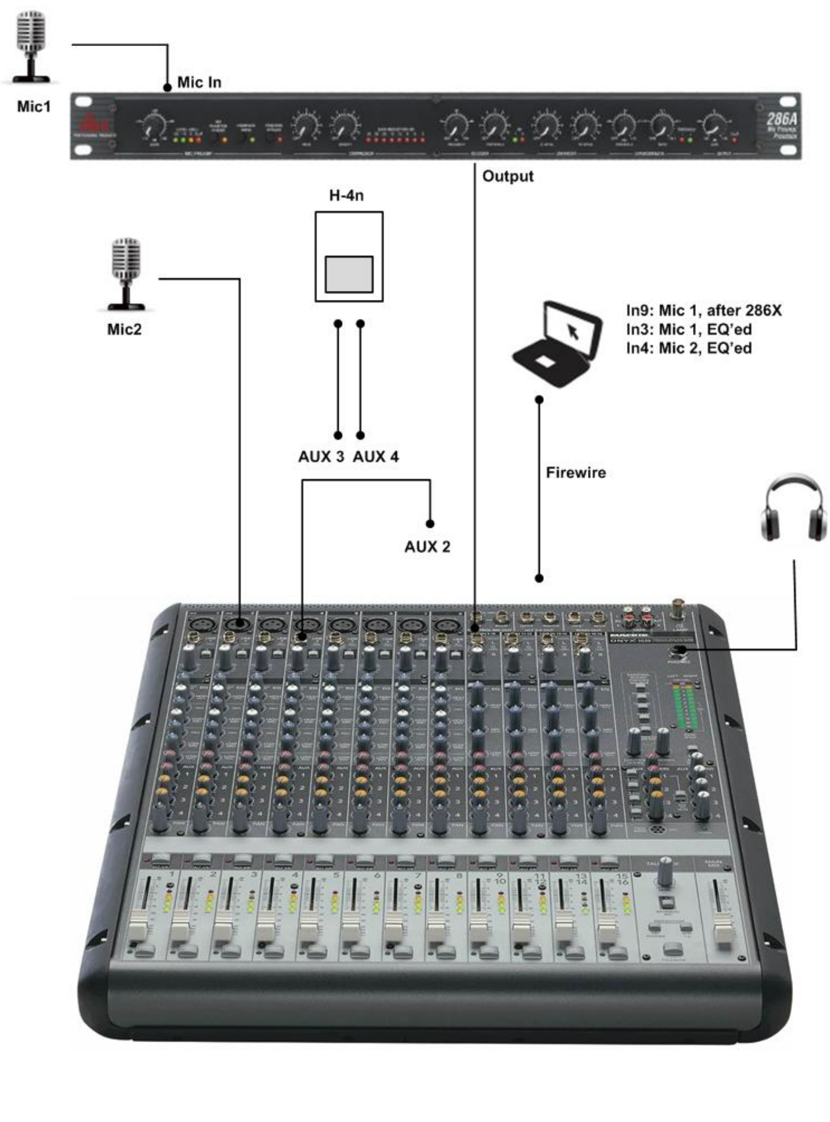
[14] http://www.conversationsnetwork.org/levelator/

[15] http://www.aphex.com/products/microphone-preamps/model-230/

Appendix 1: Wiring

The following diagrams show the actual wiring of my equipment.

Normal Setup



Telephone Recording