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Adobe Audition for Forensic Audio Analysis By Brad Barkhurst Ohio HTCIA Conference May 22, 2025





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• Have you ever taken a moment and just sat and listened to the environment you're in?





• Let's just take 30 seconds and listen.





• What did you hear?





• The fact is, these sounds are all around us and often we tune them out.





• But what if they are on a recording such as a 911 call or a cell phone video? What do we do?

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Use audio forensics! So what is it?





Audio forensics is the scientific examination, analysis, comparison and/or evaluation of audio in legal matters.

Source:

https://www.swgde.org/glossary/





What does that mean though?





What does that mean though?

It uses scientifically verified/tested tools that can be used to analyze, compare and evaluate audio recordings.

The results can be repeated and accepted in the audio forensics community.





So let's start with "sound". It's a word we hear (no pun intended) all the time. What is it, though? Can anyone define it?





Although we can't see air, let's look at vibrations that disturbs the air molecules:

YouTube: Guitar strings vibrating.



So let's imagine a pond full of calm water. Water is the medium, and it's flat. No disturbance. What happens when a rock is thrown in?



Let's watch what happens to the

water:



We have two things happen when the water medium is disturbed over time: One is the splash, which is effected by the intensity of the source (the rock).





For sound, the intensity (how loud or soft a sound is) can be measured in a unit called decibels or dB.





For human hearing:

The threshold of sound (what we can start to hear is 5 dB).

The threshold of pain (where we can get permanent hearing damage) is at 135 db.



Here is a look, at sound pressure levels in between the threshold of sound and the threshold of pain: (DAC Manual)

Notice that a typical conversation is around 65 dB.

Level (dBA SPL)	Example
/	
135	threshold of pain
125	jack hammer
115	car horn
95	subway train
75	street traffic
65	conversation
55	business office
45	living room
35	library reading room
25	bedroom at night
15	broadcast studio
5	threshold of hearing



Going back to the rock in

the water example, the other characteristic we notice is the ripples in the water. We could count how many ripples occur (frequency) within a given time.





The same thing applies for sound, the number of times (frequency) it occurs within a second can be measured. For sound the Frequency is measured in the unit of hertz (Hz). The human ear perceives this as "pitch".





The typical human ear can perceive frequencies

(dynamic range) from 20 Hz to 20,000 Hz.

To shorten things, anything over 1000 Hz are measured in Kilohertz (kHz). So the human ear dynamic range is From 20 Hz to 20 kHz.

Photo: DAC Manual

The human ear, under ideal conditions, can sense acoustic energy over the approximate range of 20 to 20,000 hertz. A hertz, abbreviated Hz, is the measurement of frequency and replaces what was once *cycles per second* (cps). Kilohertz (kHz), a unit of 1000 Hz, often is used in higher frequency measurements.

Although the human ear perceives a wide range of frequencies, it does not respond the same at all frequencies. In Figure 2.1, the Fletcher-Munson curves show contours of equal loudness level in terms of dB SPL.



Figure 2.1: Fletcher-Munson Equal Loudness Curves



So the two characteristics of sound we just mentioned are amplitude (intensity) measured in decibels (dB).

And frequency (how many times in a second) measured in hertz (Hz).



When we're working with an audio recording, it is a digital recreation of a physical (analog) sound.

So we are working with digital samples of sound(s) that occurred during a moment of time.



One characteristic of a digital recording is the sample rate. How many snapshots (samples) were made during the recording (digitization process).



In a photograph, if we are taking a photograph of someone, the photo is a sample of a moment in time.

Notice the space between the top of my head and the top of the photo. The photographer gave extra space (headroom) above my head to make sure the entirety of my face was photographed (sampled).





What happens when you don't give enough space to photograph (or sample) the head during that moment of time?



It gets cut off and you don't a have a full sample.





- The same thing applies for sound. It has a frequency rate (Hz).
- Human hearing perceives frequency up to 20,000 Hz (kHz).
- So using an audio recorder, that is sampling sound during a moment of time, you would want it to have large enough sample size that has enough headroom so you don't cut off audio frequencies.





There is a theorem called the Nyquist Theorem which, summed up, states that to get a full sample, make sure your sample is twice as much as the highest frequency you are going to sample.





So for my photo example, the photographer wants to give extra space/headroom in order to capture my entire head. If the photographer photographs an area twice the height of my head,

the photograph will definitely capture me without cutting it off.





Let's apply this to sound, to get a full digital sample, you want the recorder to have a sample rate that is at least twice high as the highest frequency.

Remember the human ear can perceive frequency up to 20,000 Hz. So to get a complete sample, you want your recording to be sampled at least 40,000 Hz. So the next time you listen to audio from an Mp3 file you downloaded, play it in VLC Media Player (the traffic cone) and select control J.

This will show you information about the file, including the sample rate. CD-quality audio has a sample rate of 44.1 kHz. (Notice it is just above 40 kHz? DVD audio is 48 kHz.)





Unfortunately, with audio we are not always provided with audio that was recorded at a 44.1 kHz (CD quality) sample rate.

Oftentimes, 911 call centers record at 8000 (8 kHz), which is much less than CD-quality?

Why? Well, cell phones don't have the frequency bandwidth of a CD-quality recording, and the software that records it also records less than CD quality, perhaps as a way to save hard drive space.

Either way, when you get a recording that has a sample rate of 8 kHz, there is only so much you can do with it because you don't have that wide range of frequencies to begin with. You cannot magically make those frequencies appear because they weren't there to begin with. When you listen to a 911 call, you'll notice that it sounds thin compared to a song on a CD.



Unfortunately, with audio we are not always provided with audio that was recorded at a 44.1 kHz (CD quality) sample rate.

Oftentimes, 911 call centers record at 8000 (8 kHz), which is much less than CD-quality?

Why? Well, cell phones don't have the frequency bandwidth of a CD-quality recording; it has a typical bandwidth of 300 – 3000 Hz, and the software that records it also records less than CD quality, perhaps as a way to save hard drive space.

When you listen to a 911 call, you'll notice that it sounds thin compared to a song on a CD.



So, what's the deal with noise?



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Noise is simply "unwanted sound".



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Typical noise in audio recordings:

Distortion (too much signal)

Reverberation (echo, jail recordings)

Electrical hum (typically around 60 Hz, AC current)

Machine noise (car engine, police siren, car AC)

Even songs playing on the radio



Noises in audio recordings can be placed into two categories:

Broadband and narrowband.

Broadband is harder to work with because it occurs all over in the frequency bandwidth often on top of what you are trying to hear. A good example of this is rain. There is not a specific single frequency for rain.



However, narrowband is easier to work with because it occurs at a specific frequency. For example, an electrical hum may just be at 60 Hz and that specific frequency can be attenuated (reduced).



Another issue in audio recordings is near/far party. Often times one person is wearing a microphone or holding a phone and you can hear them fine, but the person they are talking to is standing far away (the far party.) So it becomes a process of boosting the amplitude of the far party. However, this adds noise to the recording which often needs to be reduced.



We are typically tasked with reducing noise to increase voice intelligibility. The voice's upper limit is 5 kHz. Anything under 200 Hz is not needed for intelligibility (pg 158 DAC Manual). So, typically, a good rule of thumb is, we can automatically filter these frequencies out of the recording and keep the frequencies between 200 Hz – 5 kHz.



When working with audio, its good to have a pair of decent headphones with a good frequency response. I don't recommend noise-cancelling headphones since you won't hear the noise accurately.

Also, when cleaned up gets played in court, you'll want to have the jury listen to it on headphones or powered speakers (speakers you plug into the electric jack). Don't rely on your built-in laptop speakers.



So, enough audio theory, let's see and hear this in practice. We can use Adobe Audition to clean up audio recordings. Adobe products need to be downloaded from what's called the Adobe Creative Cloud.

https://www.adobe.com/creativecloud.html

Adobe products like Photoshop are no longer a one-time single license purchase. It is all subscription-based. There is a free month-long trial. Afterwards, it costs \$59.99 a month for all Adobe apps or \$9.99 for one app like Audition.



Once Audition is downloaded and installed, open it up. Do file open (control 0) to open an audio

recording or drag it into the interface.





We'll open up the wave file, "Raw Audio 1.wav":





What we see is a visualization of the waveform over time, of the audio recording. We can see in the lower right corner the duration, Sample rate, bit depth (number of

bits used to represent each sample), and it is recorded in stereo (left and right channel).





If we want to focus on a particular portion of the recording, if we left click in the center of the screen at the part you want to start at, holding the left mouse key down, drag to the right to select your area.

Press the space bar to play it back Or hit the blue playback icon to loop it. (cont'l + L)





To zoom in on an area, scroll your mouse wheel in or out.





You also have the area above that allows you to do quick navigation on the timeline and zoom in by

left clicking and dragging.





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left clicking and dragging.





What I typically do at this point is just to listen to the recording and hear what the issues are. So let's listen to about 30 seconds by pressing the space bar to start and stop the recording.





What do you think the big issue is in this recording?





It's a near/far party issue. Concentrate on the biggest problem first. So we want to increase the volume of the person who is speaking softly. So let's go from each place we hear that person from the

beginning.





So here I've selected the first time the far party is heard from selection time of 13 – 17 seconds (visible in the lower right selection view).





For this recording, I'm going to use the amplify tool located in the "effects" drop-down.





For this recording, I'm going to use the amplify tool located in the "effects" drop-down, select

amplitude and compression then

"amplify".





When working with audio, it's a balancing act. You don't want it overpowering and distorted but you don't want it too soft.





For this, I'll adjust my gain sliders up to increase the amplitude of the far party. Don't the levels too

high which causes distortion (red levels). Make your settings, Listen to it. Once good hit apply and then close.





For this, I'll adjust my gain sliders up to increase the amplitude of the far party. Don't the levels too

high which causes distortion (red levels). Make your settings, Listen to it. Once good hit apply and then close.



I did 15.27 dB

You should be able to see the increased waveform and hear the difference.



Before:

Before:





If you are not happy with your setting, you can hit control + U to undo it.

Keep in mind, since this is audio forensics, you will want to take screenshots of your settings so it will be reproducible. I title my screenshot settings in numerical order of use.





When you are done with this recording, go to "file", "export", "file" on the upper right corner.

Save it as a Wav file, not as an mp3 (since mp3 uses lossy compression, reducing the quality.) You also want to make sure you save it at the same sample rate as the original recording. In our case, 44.1 kHz. Change the title of your clarified audio as not to overwrite the existing audio file.





So let's open up "Raw Audio 2" and play it. (space bar)





So besides the near/far party, what's the other issue?





Hum noise is the issue. How do we figure out what frequency is the hum?





Go to the "view" dropdown and select "Show spectral frequency display"

(shift + D)





You should see something like this:





You can adjust how much waveform vs. spectral view is visible by lowering

or raising the gray line in Between the two views.





Notice how we have that bright orange color at the bottom. The more

intense the color, the louder

the sound.





Now on the right hand side you see numbers. That is your frequency

numbers (Hz).





You can use the scroll up or down on your mouse to zoom in on these

frequency numbers. But where did that intense orange color go?





So we can see that the bulk of that hum noise is under 200 Hz. So now we

have a frequency to focus on.




Go up to "Effects", "Filter and EQ" and select "30 band EQ". This brings up

the graphic equalizer (like the old school stereo systems.

We 30 bands (sliders) of frequency we can adjust.





So we can drag the frequencies under 200 Hz all the way down to -24 dB.

You can listen to it by pressing the space bar. To turn off the EQ, click on the green power button on the lower left corner of the EQ. Click it on and off to hear the before and after.





Since we also know speech intelligibility is between 200 Hz and 5 kHz, we

can reduce frequencies above

5000 hertz.





After the EQ is applied, you can visually see that the orange hum frequency has been greatly reduced under 200 Hertz.



After:





Before:

Let's say you have multiple tones at specific frequencies. You want to focus in on those specific frequencies and reduce them. Open up "Raw Audio 3.wav" and look at it using the spectral view. How many tones do you think there are in this recording?



Let's say you have multiple tones at specific frequencies. You want to focus in on those specific frequencies and reduce them. Open up "Raw Audio 3.wav" and look at it using the spectral view. How many tones do you think there are in this recording?

Answer: 3



Now, what frequencies are the tones occurring at?



Now, what frequencies are the tones occurring at? Once you have an idea, write the frequencies down.





We could use the graphic EQ to reduce the frequencies, but there is a tool that is even more precise!





Go to "Effects" dropdown, "Filter and EQ" and select the "Notch Filter".





Go to "Effects" dropdown, "Filter and EQ" and select the "Notch Filter".





These notches can be used to reduce specific frequencies by reducing it's specific gain (amplitude).

By default, these notches are already active, so let's disable them by clicking on each number to the right of the word "Enable".





When deselected, it should look like this:





With the notch filter not enabled press the space bar and you can see the tonal noise. Now looking at the lowest in the noise, where do see the peak?





With the notch filter not enabled press the space bar and you can see the tonal noise. Now looking at the lowest in the noise, where do see the peak? Answer: 60 Hz

So let's reduce it by enabling notch filter 1 and typing in 60 Hz. Then press the space bar to play the recording with the 60 Hz notch.





See how it got lowered?

Before:



After:





The other thing you can do is left click and drag that white #1 left or right if you feel the frequency you typed in is a little. The other thing you can do is left-click on the white circle above the one and drag down. This increases how

much reduction gain is applied. Pretty cool huh?

Now repeat this process with the other the tonal frequencies, which are?



These EQ filter tools are nice but they will not work on noise that changes pitch throughout the recording such as a police siren where the frequency curves up and down like a sine wave. There are tools out there that allow you to visualize that sound in a spectral view, then trace over it, and cancel it out. It's a process called spectral noise cancellation. A tool like Spectral Layers Pro can do this, even reducing harmonic noise.





There are a lot more audio tools out there, such as Izotope RX, but that's all we have time for today.

For best practices on audio forensics, visit SWDGE.org

Also check out the audio forensics training from LEVA.org and Salient Sciences.com (Formally Digital Audio Corporation.)

Feel free to keep practicing on the audio files and show me what you come up with!



QUESTIONS?

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