Supplement to Chapter 4 of *The Science of Digital Media* – Digital Audio Processing

Worksheet – Digital Audio > Introduction to Digital Audio Processing Programs[1]

This worksheet can be done with a digital audio processing program like Audition, Sound Forge, or Jokosher.

Sampling Rate and Aliasing

Begin a new audio file. Choose a sampling rate of 44,100 KHz, a bit depth of 16 bit bits, and mono.

Generate a single-frequency sine wave tone at 2100 Hz.

Zoom in to see the sine wave. (Look for a zoom tool. It will probably have a magnifying glass icon.)

Play the file.

Display the Frequency Analysis View.

Question 1: (a) Describe what the Frequency Analysis window shows.

Assuming your audio processing program has a multitrack view, put the tone you just generated into one of the tracks of the multitrack view.

Create another waveform – same sampling rate, bit depth, and mono.

Generate a tone as before, but this time at a frequency of 480 Hz. Insert this tone into the multitrack view.

Combine the two tracks into one waveform. This is called "mixing down."

Switch back to the waveform view. Look at the wave form. Zoom in if necessary.

Question 2: Why does the wave look different? Explain what mixing down does to the two waves, in mathematical terms.

Convert to a sampling rate of 3100 samples. Do not change the bit depth; keep it at 16 bits. If there is an option for pre/post filter, deselect it.

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Now play your file again.

Question 3: (a) In what way does the audio file sound different? Can you still hear both tones (i.e., are both wavelengths still present)? Explain. (b) What phenomenon is being illustrated here? (c) Explain, by means of the Nyquist theorem, how the frequency components of this audio clip have changed.

Quantization, Resolution, Quantization Error, and Dithering

Open a WAV file for a clip of a piece of classical music (or whatever file your lab instructor gives you). Use just a short clip – preferably one that has a big difference between the loudest and softest parts. (You should be able to find sample WAV files on the web.) Look at the waveform of sound file, and play it.

Look at the spectral view of the waveform.

Question 4: Explain how you read the spectral view and describe what it tells you about this clip of music.

To understand the effects of resolution on sound quality, let’s choose an unrealistic bit depth that will exaggerate the quantization error. We'll requantize the file at this bit depth.

Reduce the number of bits per sample to 6 (as opposed to the original 16). Be sure NOT to enable dithering. Do NOT change the sample rate, just the resolution.

Question 5: Look at the waveform of your altered sound file.
(a) Describe what the waveform looks like now. (b) Why does it look like this? Be specific, giving a mathematical explanation.

Question 6: Go back to the spectral view. Describe how it looks and explain it.

Play the file.

Question 7: (a) What is different about the way the music sounds now? (b) What is the cause for this difference? Be specific, giving a mathematical explanation in terms of waveforms.

Undo the bit depth conversion that you did. Then convert the sample type to 6 bit samples again, but this time, enable dithering. If you have a choice, you can choose the triangular probability distribution function and dither at a depth of 1 bit.

Question 8: Describe how the wave now looks in the spectral view and why.

Play the file.

Question 9: (a) What's the difference in how the music sounds now, compared to how it sounded without dithering? Be specific, particularly in describing the low-amplitude parts of the sound file. (b) What does the file look like now, compared to the first time you converted the sample type without dithering? (c) How does dithering accomplish this effect, and how can it help improve the quality of the sound? Be specific.

Leave this file open for the next part.

Decibels and Dynamic Range

Question 10: Tell what classical music piece you've been using for your WAV file. Would you say that the audio file has a wide or narrow dynamic range? (This is a judgement call, but explain your answer.)

Question 11: Question 10 asks about the dynamic range of the piece of music. We can also speak of the dynamic range of a file on the basis of its bit depth. (a) What was the dynamic range of the file when we still had it at a resolution of 16 bits? (b) What is its dynamic range after you convert it to 6 bits? Explain how you know the answers to a and b.