Digital Asset Development: Lab Session 5 – Audio Processing and Adobe Premiere Pro CS6

Introduction

This lab session is divided into two parts. The first continues where we left off last week, looking at some more advanced audio processing concepts in Audacity, with reference to the ideas covered in the lectures. The final part forms an introduction to Adobe Premiere Pro, with a focus on how audio clips can be organised within the package. This serves as a prelude to next week's class, where we will explore Premiere in much more detail.

Part 1: Audio Processing in Audacity

In these exercises we will be investigating the tools Audacity gives us for analyzing and processing audio using frequency information. This continues the themes covered in today's lecture. Although the concepts covered here may seem a bit theoretical, they can be directly applied when cleaning up sounds for use as project assets.

In an empty Audacity window, go to the Generate menu and choose Tone. This command allows very simple waveforms to be created. The popup dialogue allows you to define the basic shape of the wave, as well as its frequency, amplitude and duration. Choose a Sine waveform and leave the settings at their defaults, apart from the duration which you should set to 5 seconds. Click OK. When you listen to the resulting waveform you will hear a continuous beep – this is what a pure tone at a single frequency sounds like. If you zoom into the clip you will see the wave shape below.

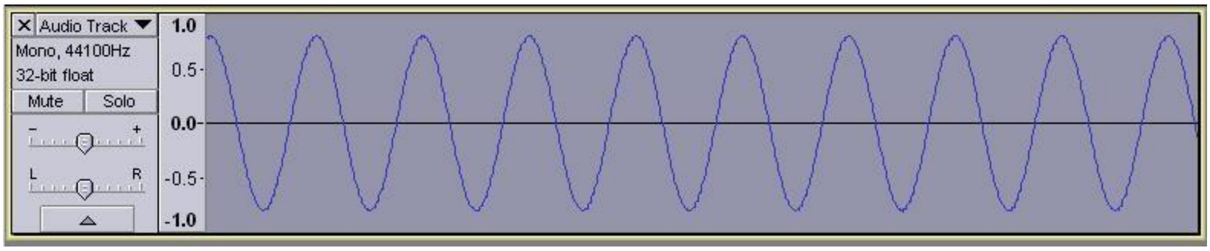

Fig 1: waveform for a single tone (sine wave)

If you switch to spectrogram view for the track (using the dropdown menu in the track header), you will observe that the energy for the clip is practically all contained in the white bar centered on the wave's fundamental frequency of 440 Hz.

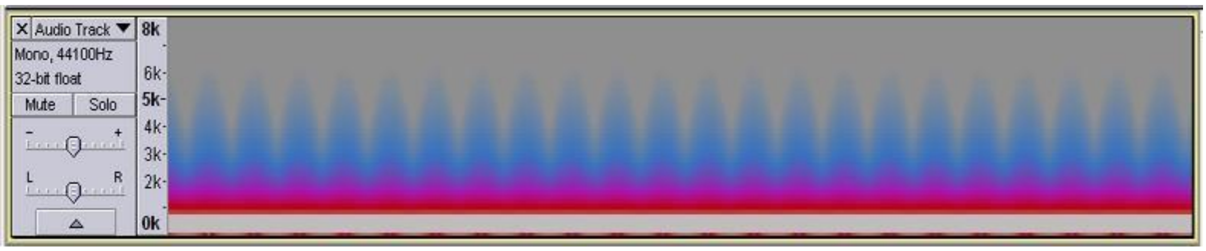

Fig 2: spectrogram of the sine wave

Go back to the waveform view, and choose Effect > Amplify. This function adjusts the amplitude, and hence the volume, of an audio waveform. Dragging the slider left decreases the amplification (measured in dB). You may notice that when you drag to the right, once the new peak amplitude goes above 0 the OK and Preview buttons are disabled. To enable them again, tick the Allow Clipping checkbox. Position the slider so the new peak amplitude is around 10 dB, and click OK. The resulting wave should look something like the screenshot below.

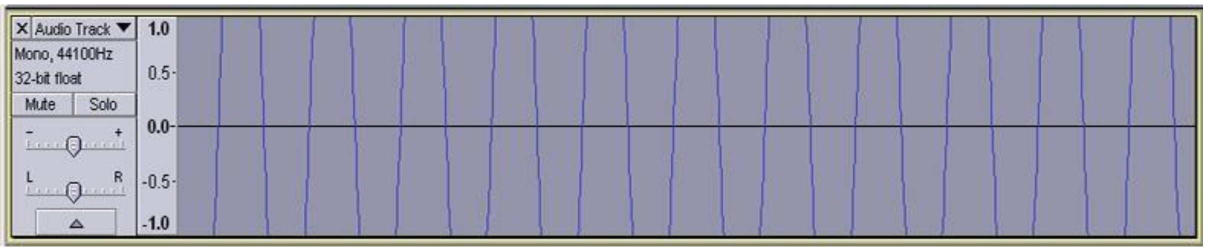

Fig 3: result of amplifying the sine wave

Clearly, we have managed to chop off the extremities of the waveform by trying to amplify the sound beyond the available dynamic range. Play the track to hear the effect this has on the sound itself. You should note that, although the basic pitch is the same, it has a high buzzing overtone. This is because we have introduced additional frequencies to the clip above our original base frequency.

Undo the Amplify command to return to the original waveform. Add a new track to the workspace (use Tracks > Add New > Mono Track), and with this track active choose Generate > Tone once more. This time create a square wave with the other settings left the same as before. This gives a wave in the form shown below. Playing the resulting sound back shows an even stronger and more unpleasant buzz is present in the track (note: you should mute the track containing the original waveform so as to hear just the square wave).

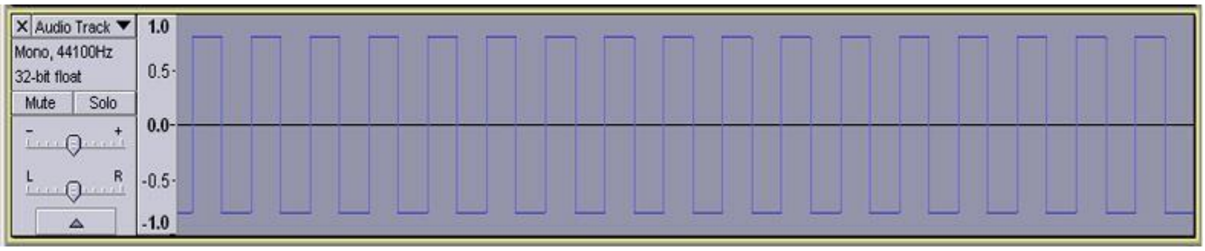

Fig 4: square wave form

Now switch the square wave to spectrogram view. The result looks quite different from the sine wave (Fig 2), with additional bars appearing at regular intervals on the plot.

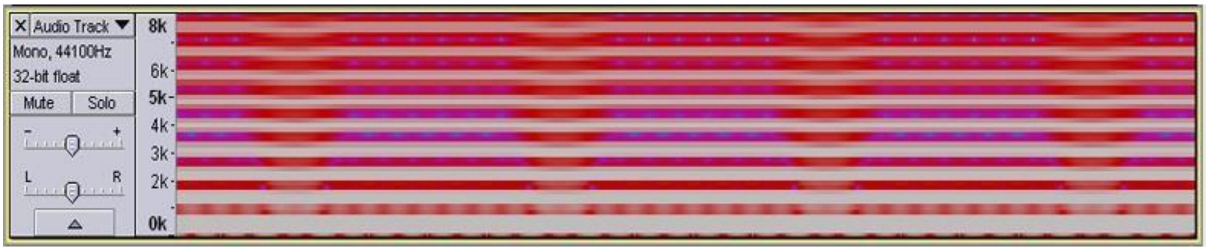

Fig 5: spectrogram of square wave

These bars show the contributions to the overall sound by the higher frequencies that produce the buzzing noise. It is hard to work out from the scale on the spectrogram plot, but the spacing of the bars is defined by the base frequency of the original square wave. To see this more clearly we can use the Plot Spectrum function to generate the frequency spectrum of each wave.

Click in the track header of the sine wave and choose Analyze > Plot Spectrum. This calculates the Fourier Transform of the wave, which should look like the left side of the picture below. The scale is slightly confusing, as the 86 Hz label refers to the very left hand edge of the graph. You can verify that the purple peak lies on the wave frequency of 440 Hz by moving the mouse over the graph and reading the output from the field labeled *cursor*. Close the spectrum window, click in the square wave track header and do Analyze > Plot Spectrum once more. The new plot should look like the right side of Fig 6.

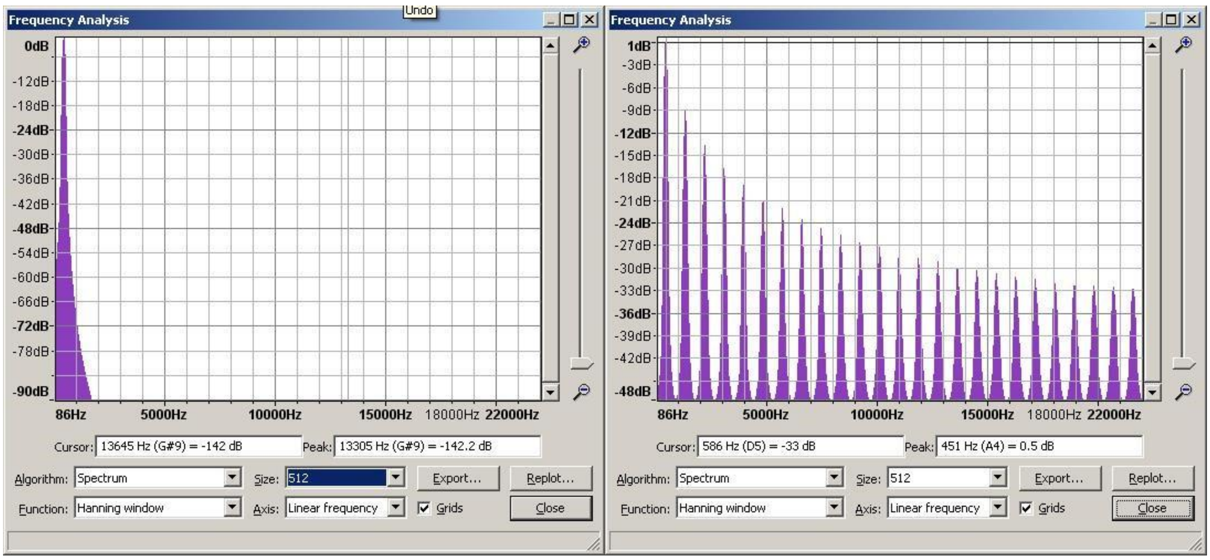

Fig 6: frequency spectrum graphs for (left) sine wave, and (right) square wave of 440 Hz

The series of peaks on the square wave graph correspond to the bars on the spectrogram. If you use the cursor to inspect them you will find the second, third and fourth peaks appear at 1320, 2200 and 3080 Hz respectively. These values are 3, 5 and 7 times the base frequency of 440 Hz, indicating that the extra peaks are *harmonics* of the original tone. This is why, although the buzzing is annoying, the sound is not actually discordant.

We are now going to apply a basic filter to part of the sound to remove the additional harmonics and effectively convert the square wave into a sine wave. To do this, we need to be able to select a specific time/frequency segment of the spectrogram. In the track header dropdown for the square wave, choose Spectral Selection. This looks just like the spectrogram view, but it is now possible to make the appropriate selection by clicking and dragging the plot. Zoom out so the whole wave is in view. Then position the cursor partway along the clip and just above the lowest of the white bars, and drag to the right. The distance dragged defines the time span of your selection. Then drag up to the top of the clip to define your selection rectangle, the extent of which should be highlighted in yellow as shown below.

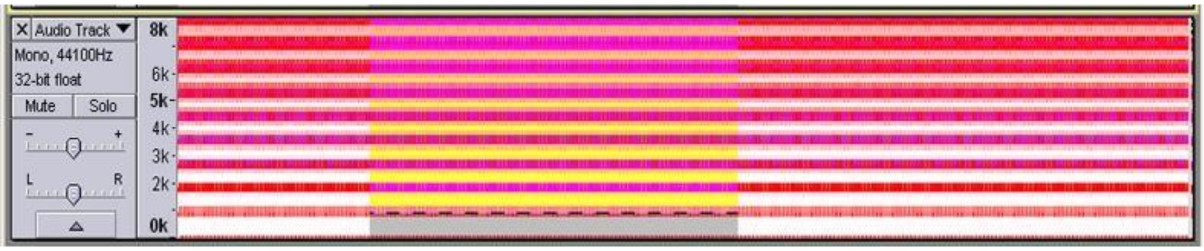

Fig 7: making a spectral selection

With your selection made, go to Effect > Low Pass Filter (it is near the bottom of the list). Set the cutoff frequency to 700 Hz and rolloff to the maximum 48dB. This will remove the frequencies above the cutoff level. The resulting spectrogram should now look like Fig 8.

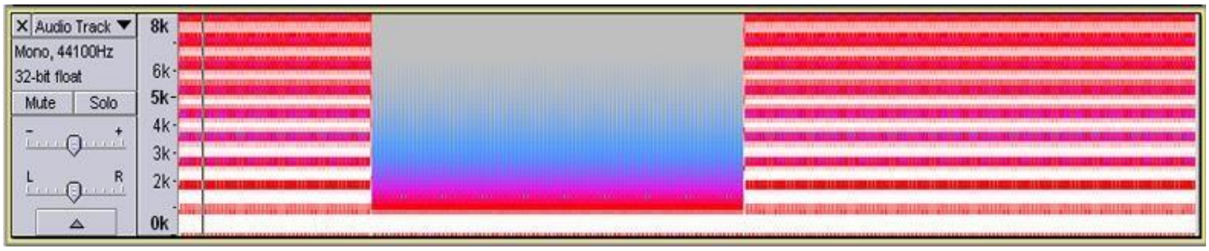

Fig 8: result of low pass filter on a square wave

Playing the full clip back, you should find the buzzing has largely been removed leaving just the basic tone. The Plot Spectrum function shows that the frequency distribution is now the same as for a sine wave of 440 Hz, indicating that we have successfully removed the extra harmonics.

The *low pass* filter suppresses high frequencies above a defined cutoff threshold. We can use a *high pass* filter if we want to remove low frequencies from a sound. Take the same approach as before, using the high pass option from the Effects menu, to selectively remove the base 440 Hz tone from another section of the square wave clip, leaving just the buzzing harmonics.

Another option from the Effect menu that can be used within the spectral selection track view is the Spectral Edit Multi Tool. This is rather tricky to use – basically, you need to define a rectangle in the spectrogram, select the tool from the menu, and then play the sound back to see if it has done what you wanted. There is some guidance available in the Audacity manual in part 4 of the Spectral Selection chapter.

You may be wondering why you need to worry about editing sine waves and square waves, as you wouldn't want to listen to them anyway. However, these methods are standard techniques for fixing audio issues for music or speech recordings. Any recording may need its frequency balance altered – this is what a graphic equalizer in a stereo system does. Some recordings are affected by noise at specific frequencies, often due to electrical interference, and spectral methods offer an ideal solution. The appearance of extra harmonics is also common when recording on cheap equipment or in uncontrolled conditions outside of a studio. Understanding and using these methods will give you more options for ensuring your sound assets are of good quality.

Click Removal

We can also use Audacity's click removal tool as a means of improving audio recordings. Download the file *clicky.wav* from Moodle and bring it into Audacity. It is a short clip of a voiceover from a video tutorial, which suffers from a couple of clicks. These are a consequence of recording via a headset, and should be visible as spikes in the waveform at around 1.8 and 2.7 seconds.

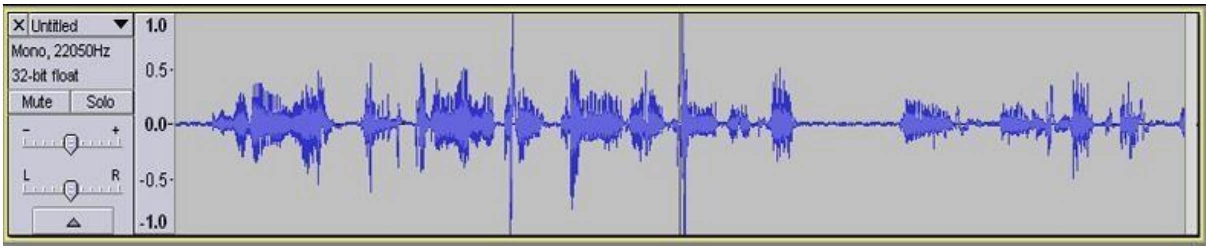

Fig 9: speech sample with two clicks

Zoom in on the second of the clicks and switch to spectrogram view. Unfortunately, in this case the spectrogram is not much use, as it is hard to see any difference between the clicks and the general speech sounds. Go back to waveform view and select the part of the wave containing the click. Then choose Effect > Click Removal. The dialogue window has two settings – Threshold and Max Spike Width. Try using the default settings first – if you get an error message about needing more samples then you'll need to redo your selection. To get the required number of samples, you will need around 0.2 seconds of audio. Once the tool accepts the selection, try experimenting with the values to see what works. The shot below shows the effect of using a threshold of 50 and spike width of 30 on the selection shown.

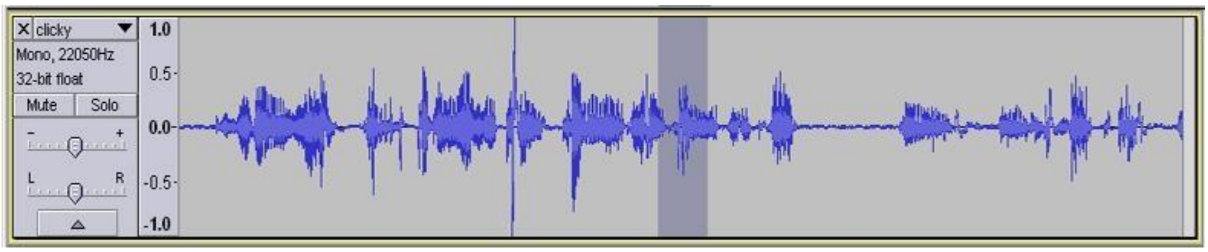

Fig 10: speech sample with click 2 removed

Use the click removal tool to eliminate the other click. Will using the same settings work in the same way?

Part 2: Introduction to Adobe Premiere Pro CS6

Adobe Premiere Pro CS6 is a fairly sophisticated video editing tool that forms part of Creative Suite/Creative Cloud. It is a nonlinear editor that enables a range of editing tasks including cutting together video (and audio) clips, adding transitions and titles, and applying a variety of filters and effects. One of Premiere's strong points is its integration with other Adobe tools, in particular Photoshop and After Effects.

The Premiere Interface

Like all Creative Suite applications, Premiere's interface is based around a series of panels that are typically arranged into thematic groups. The screenshot below shows the usual default layout for Premiere Pro CS6. The identity and functions of each panel, reading from left to right and top to bottom, are summarized in the table.

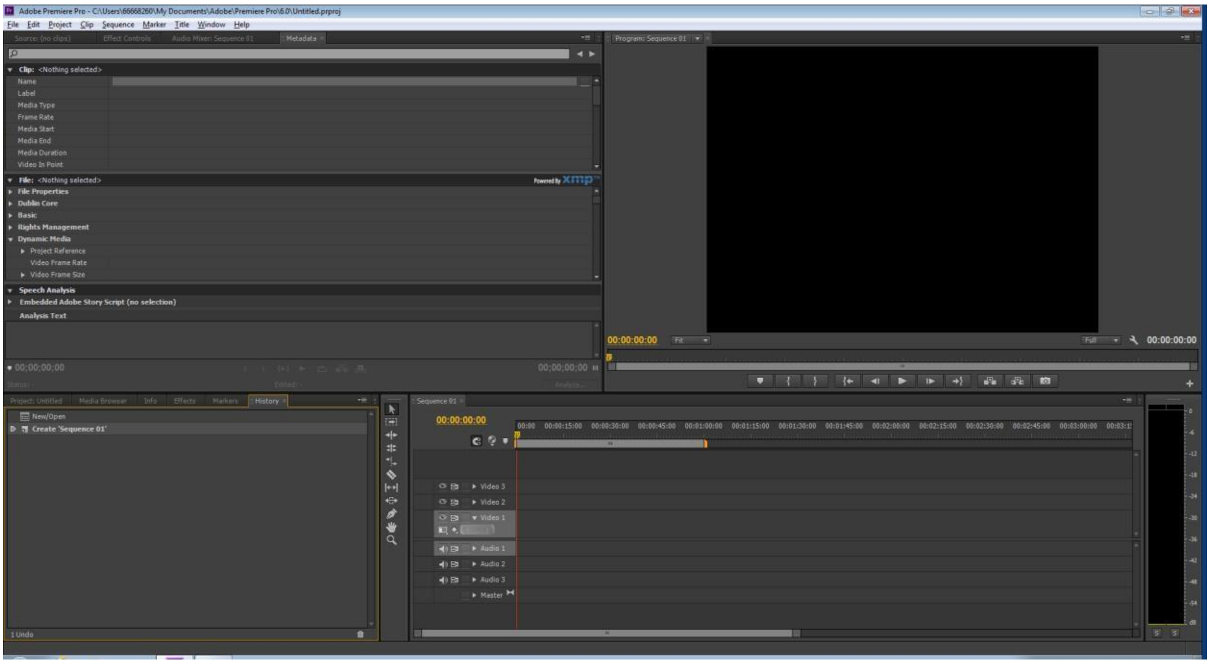

Fig 11: The Adobe Premiere Pro CS6 interface

Source Monitor	Allows playback and preview of individual video and audio clips
Effects Controls	For editing effects applied to a selected clip
Audio Mixer	Controls audio settings and allows direct recording of audio
Metadata Panel	Lists metadata associated with a file
Program Monitor	Allows playback and preview of the timeline contents
Project Panel	Lists the audio, video and image files and other project assets
Media Browser	Allows browsing of files on the hard drive
Info Panel	Provides context sensitive information on the selected item
Effects Panel	Contains video and audio effects and transitions
Marker Panel	Shows the marker points in a clip or sequence
History Panel	List of recent actions

Tools Panel	Gives access to the different tools available in Premiere
Timeline	Place where assets are assembled into a sequence
Audio Meters	Volume monitor for sound (active when previewing the timeline)

The tools panel contains 11 tools that offer different aspects of Premiere functionality. These are listed below.

Selection tool	Usual default tool; allows selection of items or interface elements
Track selection tool	Allows selection on the timeline of all clips to the right of the cursor
Ripple edit tool	Allows trimming of a timeline clip preserving surrounding edits
Rolling edit tool	Allows editing of a pair of clips while preserving overall duration
Rate stretch tool	Speeds up or slows down a clip in the timeline
Razor tool	Splits a clip in the timeline
Slip tool	Alters a clip's in and out points while preserving its duration
Slide tool	"Slides" a clip along the timeline while retrimming surrounding clips
Pen tool	Sets and selects keyframes for animation of clip properties
Hand tool	Allows scrolling through the timeline
Zoom tool	Enables zooming within the timeline

Audio in Premiere
This week, you will simply learn about loading, playing and arranging audio clips on the timeline in Premiere. You can use the files *QMB.wav* and *DAD.wav* from Moodle as example clips. Open up Premiere Pro from the CS6 menu. When prompted, start a new project using the default settings. Use File > Import to load in your audio clips, which should then be visible in the project panel. Double-clicking the listing should bring up a waveform in the source panel. You can use the playback controls to preview the clip.

You should view the following YouTube tutorials to get started with using audio in Premiere.

- https://www.youtube.com/watch?v=_YuMKP2DPfw Premiere Pro CS6: Working with Audio – this is a Lynda.com tutorial that looks at stereo and mono audio tracks in Premiere, and how the software deals with them; it also covers separating an audio soundtrack from a video file.
- https://www.youtube.com/watch?v=qwbUX8MoXK4 Premiere Pro: Using Keyframes to Control Audio Volume – this gives a detailed, if rather long-winded, overview of volume editing control options in Premiere; note that the tutorial uses CS5 instead of CS6 so some interface elements are slightly different.

Having viewed these videos, create a simple audio sequence in the timeline, using *QMB.wav* as a backing track with *DAD.wav* overlaying it part way through. The backing should fade down when the vocal sample appears and return when it has finished. Note that one of the tracks is in mono form while the other is in stereo.