CHAPTER 8

The Algorithms of Digital Audio: Audio Processing

Now that you have an understanding of the hands-on digital audio editing tools in Audacity, it's time to look at how you can use **algorithms**, which take the form of **plug-in filters** and **effects**, to apply waveform editing operations and special effects with math and code routines.

You'll look at a number of the primary audio editing and "sweetening" effects in the Audacity 2.1 Effect menu. Some of them visualize the concepts that you learned about in the first few chapters of the book. This offers reinforcement of the principles. You will learn how to utilize algorithms that allow you to amplify waveforms, shift the pitch, change the playback speed, equalize the frequency response, and apply audio effects such as reverb and echo (delay). You also learn how to selectively filter frequencies out of your sample waveform by using audio filters such as High Pass, Low Pass, and Notch Filters.

Audacity 2.1 has 44 default Effect menu entries, so I can't cover them all in this chapter. I'll cover the ones that demonstrate the concepts you have already learned, and some of the others that are editing mainstays; so you'll learn all the basic processing.

It's important to note that there are another 80 plug-in filters that you can add to the Effect menu by using **Effect ➤ Manage**, shown at the top of the Effect menu in Figure 8-1.

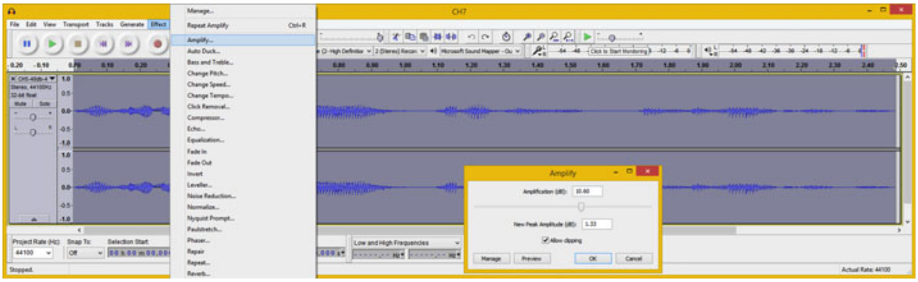

Figure 8-1. *Select sample and use an Effect ➤ Amplify algorithm*

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Algorithmic Audio Effects Processing

Algorithms are code (usually C++ in digital audio editing) that process your sample waveform. This is why it is important that you use a high-quality sampling frequency—at least CD quality (44.1 kHz) or THX quality (48 kHz)—and leave the sample resolution at 32 bits until the editing and processing is done and you are ready to perform data footprint optimization.

The reason for using the highest possible sampling frequency and sample resolution is that you want to give as much data to the algorithm as you possibly can. The more data the algorithm has, the better results it can provide. Later, after all the processing is complete, you can reduce your sample resolution to 24-bit (HD), or more likely to 16-bit, which is CD quality. You can preview using your ears, your eyes (waveform), and even a data result (export audio as PCM or FLAC), as you've seen in previous chapters and will see in this chapter.

Waveform Amplitude: The Amplify Effect

The first effect that you apply increases the y-axis dimension in the amplitude of the waveform. The result of this is amplification, more commonly referred to as "turning up the volume." Open your **CH7.aud** Audacity project and select the entire waveform, as is shown in Figure 8-1. Select **Effect ➤ Amplify** and enter an amplification factor in decibels; I used 10.6, which is a 133% peak amplitude (a 1.33 factor). Next, select the **Allow clipping** option. You can also use the Preview button to preview different settings until you find one that sounds good to you.

Once you click the **OK** button to apply an Amplify effect, you are able to see this effect algorithmically applied to your waveform. This is shown in Figure 8-2, and as you can see, the waveform looks (and sounds) drastically different.

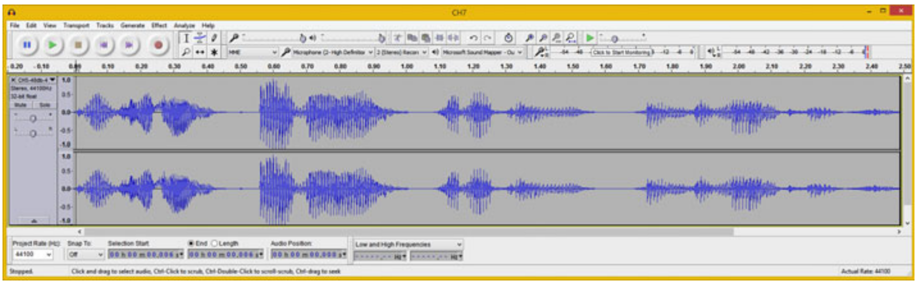

Figure 8-2. *Amplitude y-axis dimension of waveform is magnified*

The sample is indeed louder on playback; however, as you can hear, the quality is lower. I don't hear any artifact introduction, but the clarity of the vocals has suffered, so I used the **Edit ➤ Undo Amplify** menu sequence to undo the algorithm processing. I will tell my app users to turn up their volume.

If you look at the amplified data sample waveform shown in Figure 8-2, it is obvious that this sample waveform has been stretched (only) along the y axis, which is the **amplitude** (volume) of your sample. This effect reinforces what you learned about audio sample characteristics in Chapter 3.

Next, let's take a look at the x axis, or frequency, of the waveform. Changing the frequency of your waveform is called **pitch shifting**.

Waveform Frequency: The Pitch Shifting Effect

Next, let's take a look at the other sample characteristic that you learned about in Chapter 3, sample frequency. To change your data sample frequency, you use the **Effect ➤ Change Pitch** menu sequence, which is shown on the left side of Figure 8-3. I shifted the pitch up one octave, or 100%, so that my vocal sounds like a member of the popular chipmunk trio.

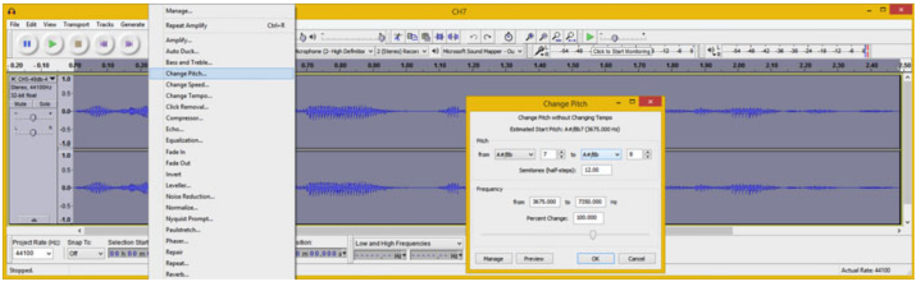

Figure 8-3. *Select sample, then Effect ➤ Change Pitch algorithm*

As you can see in the **Change Pitch** dialog, there are 12 semitones to an octave, and the **Percent Change** data field shows that there is a 100% full octave pitch increase indicated.

If you are sampling notes using an instrument that is in tune, you can also select pitch changes by notes. I clicked the **OK** button to apply the algorithm. This time, the x axis of the sampled waveform was affected and the amount of data was doubled in this dimension, creating a waveform twice as dense, as you can see in Figure 8-4.

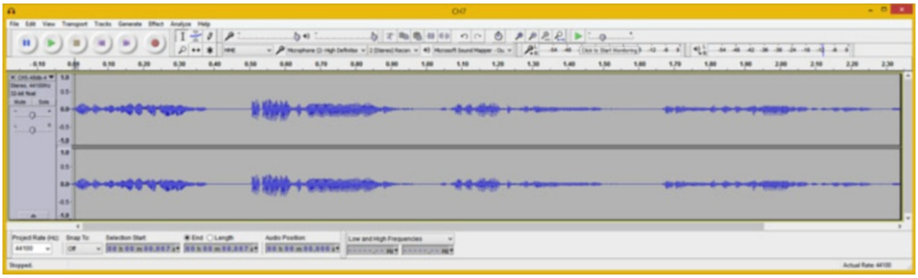

Figure 8-4. *Frequency x-axis dimension of sample is compressed*

Since I'm not currently working on a chipmunks movie or games, I used the **Edit ➤ Undo Change Pitch** menu sequence to return to the original data sample waveform.

Waveform Speed: Vinyl Record Playback Speeds

An algorithm similar to the Change Pitch effect is the **Change Speed** effect (see Figure 8-5). It does much the same thing to your frequency (the x axis) without keeping the sample length (time) the same. What this does is shorten the (time) length for your sample. Interestingly, the paradigm that the Change Speed dialog uses is old-fashioned vinyl record RPMs! The baseline is large 33.3 RPM records with options to change the playback speed to match a 45 RMP record, or even an ancient 78 RPM record. As you can see, the dialog also allows you to specify speed adjustment using a multiplier, percentage, or actual time value, all of which are calculated in real time.

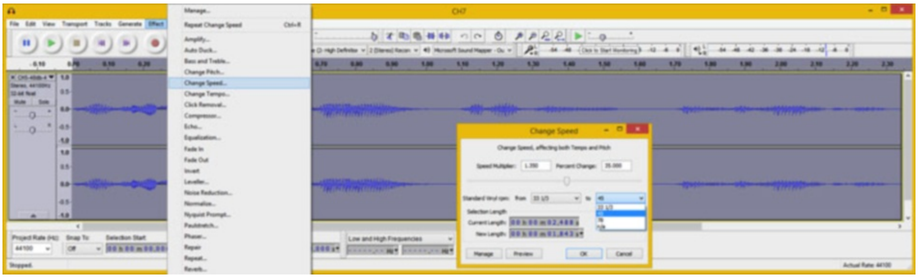

Figure 8-5. *Select sample, then Effect ➤ Change Speed algorithm*

As you can see in Figure 8-6, your data sample waveforms are compressed along the x axis like they were in Change Pitch, but the sample is shorter (rather than length being maintained), as you can see on the right. This 35% change is the amount of gray shown on the right in Figure 8-6; you can see the multiplier and percent change numbers that you specified in the dialog visually inside the Audacity 2.1.1 application.

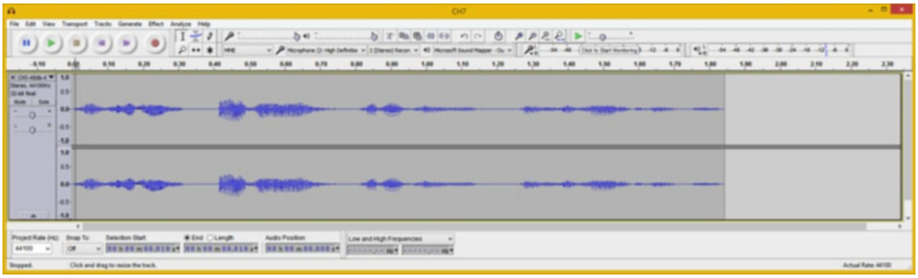

Figure 8-6. *Frequency x axis and length of sample is compressed*