5 Chapter 5 Digitization

5.1 Concepts

5.1.1 Analog Vs. Digital

Today's world of sound processing is quite different from what it was just a few decades ago. A large portion of sound processing is now done by digital devices and software – mixers, dynamics processors, equalizers, and a whole host of tools that previously existed only as analog hardware. This is not to say, however, that in all stages – from capturing to playing – sound is now treated digitally. As it is captured, processed, and played back, an audio signal can be transformed numerous times from analog-to-digital or digital-to-analog. A typical scenario for

This material is based on work supported by the National Science Foundation under CCLI Grant DUE 0717743, Jennifer Burg PI, Jason Romney, Co-PI.

live sound reinforcement is pictured in Figure 5.1. In this setup, a microphone – an analog device – detects continuously changing air pressure, records this as analog voltage, and sends the information down a wire to a digital mixing console. Although the mixing console is a digital device, the first circuit within the console that the audio signal encounters is an analog preamplifier. The preamplifier amplifies the voltage of the audio signal from the microphone before passing it on to an **analog-to-digital converter** (**ADC**). The ADC performs a process called **digitization** and then passes the signal into one of many digital audio streams in the mixing console. The mixing console applies signal-specific processing such as equalization and reverberation, and then it mixes and routes the signal together with other incoming signals to an output connector. Usually this output is analog, so the signal passes through a **digital-to-analog converter** (**DAC**) before being sent out. That signal might then be sent to a digital system processor responsible for applying frequency, delay, and dynamics processing for the entire sound system and distributing that signal to several outputs. The signal is similarly converted to digital on the way into this processor via an ADC, and then back through a DAC to analog on the way out. The analog signals are then sent to analog power amplifiers before they are sent to a loudspeaker, which converts the audio signal back into a sound wave in the air.

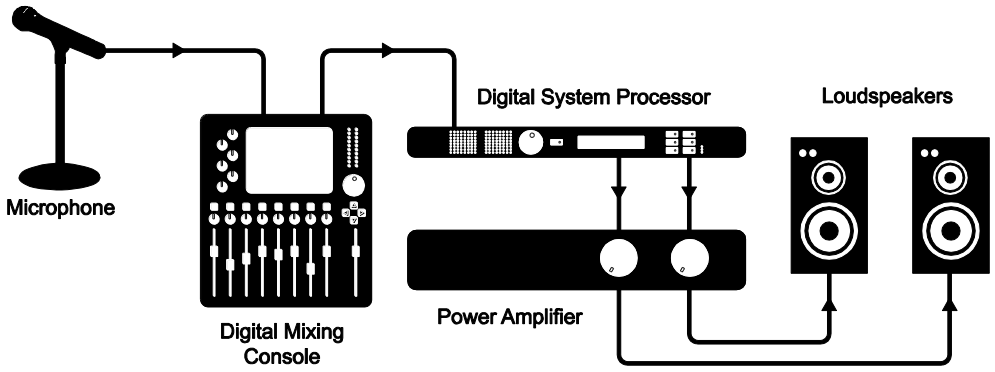


Figure 5.1 Example of a simple live sound reinforcement system

A sound system like the one pictured can be a mix of analog and digital devices, and it is not always safe to assume a particular piece of gear can, will, or should be one type or the other. Even some power amplifiers nowadays have a digital signal stage that may require conversion from an analog input signal. Of course, with the prevalence of digital equipment and computerized systems, it is likely that an audio signal will exist digitally at some point in its lifetime. In systems using multiple digital devices, there are also ways of interfacing two pieces of equipment using digital signal connections that can maintain the audio in its digital form and eliminate unnecessary analog-to-digital and digital-to-analog conversions. Specific types of digital signal connections and related issues in connecting digital devices are discussed later in this chapter.

The previous figure shows a live sound system setup. A typical setup of a computer-based audio recording and editing system is pictured in Figure 5.2. While this workstation is essentially digital, the DAW, like the live sound system, includes analog components such as microphones and loudspeakers.

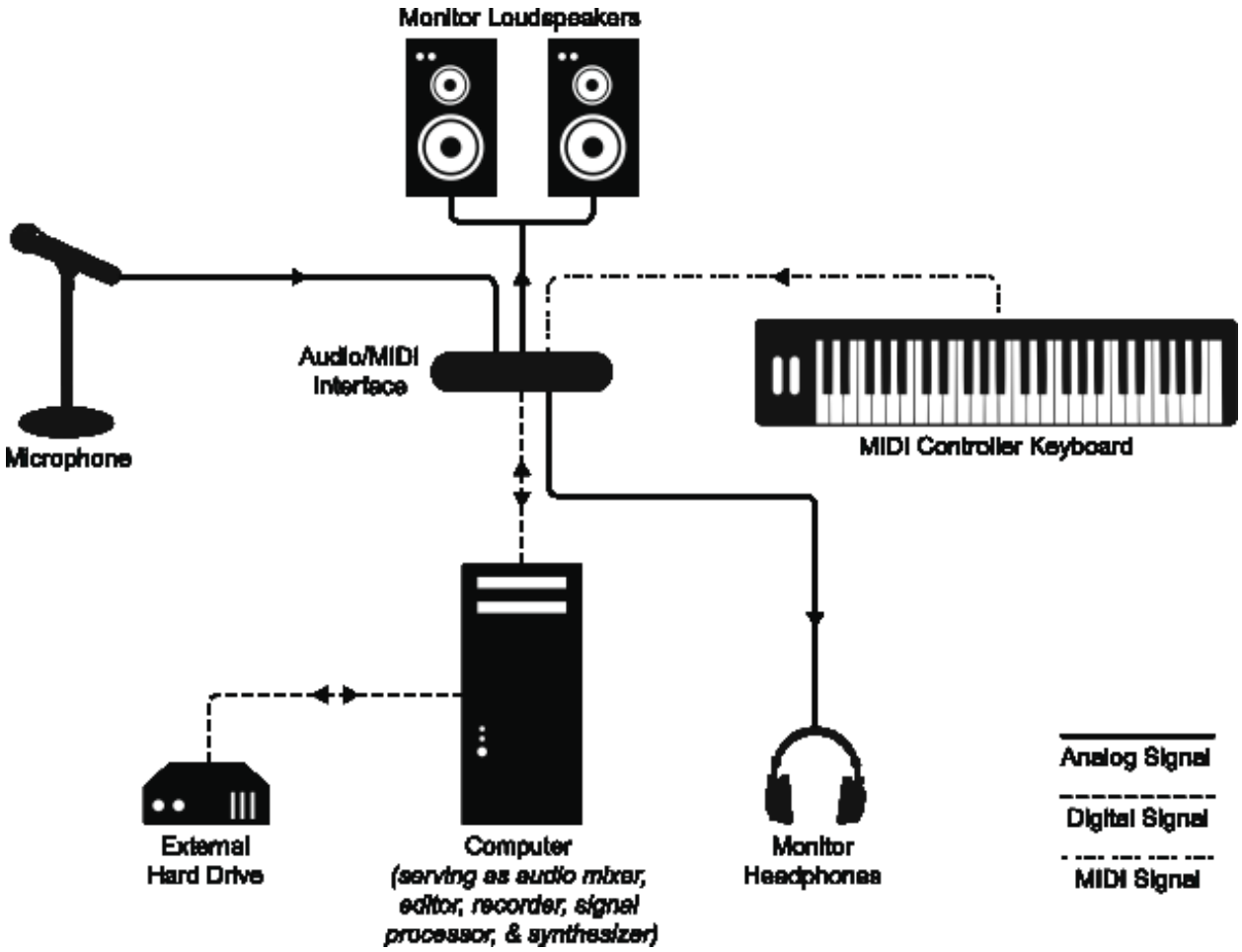


Figure 5.2 Analog and digital components in a DAW

Understanding the digitization process paves the way for understanding the many ways that digitized sound can be manipulated. Let's look at this more closely.

5.1.2 Digitization

5.1.2.1 Two Steps: Sampling and Quantization

In the realm of sound, the digitization process takes an analog occurrence of sound, records it as a sequence of discrete events, and encodes it in the binary language of computers. Digitization involves two main steps, sampling and quantization.

Sampling is a matter of measuring air pressure amplitude at equally-spaced moments in time, where each measurement constitutes a **sample**. The number of samples taken per second (samples/s) is the **sampling rate**. Units of samples/s are also referred to as **Hertz** (Hz). (Recall that Hertz is also used to mean cycles/s with regard to a frequency component of sound. Hertz is an overloaded term, having different meanings depending on where it is used, but the context makes the meaning clear.)

Quantization is a matter of representing the amplitude of individual samples as integers expressed in binary. The fact that integers are used forces the samples to be measured in a finite

number of discrete levels. The range of the integers possible is determined by the **bit depth**, the number of bits used per sample. A sample's amplitude must be rounded to the nearest of the allowable discrete levels, which introduces error in the digitization process.

When sound is recorded in digital format, a sampling rate and a bit depth are specified. Often there are default audio settings in your computing environment, or you may be prompted for initial settings, as shown in Figure 5.3. The number of channels must also be specified – mono for one channel and stereo for two. (More channels are possible in the final production, e.g., 5.1 surround.)

Aside: It's possible to use real numbers instead of integers to represent sample values in the computer, but that doesn't get rid of the basic problem of quantization. Although a wide range of samples values can be represented with real numbers, there is still only a *finite* number of them, so rounding is still be necessary with real numbers.

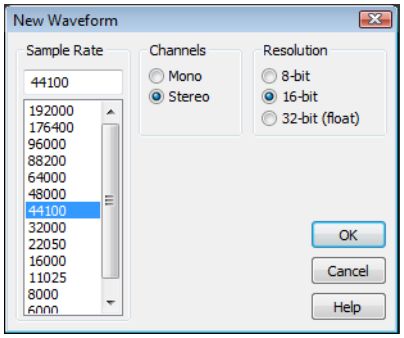


Figure 5.3 Prompting for sampling rate, bit depth, and number of channels

A common default setting is designated **CD quality audio**, with a sampling rate of 44,100 Hz, a bit depth of 16 bits (i.e., two bytes) per channel, with two channels. Sampling rate and bit depth has an impact on the quality of a recording. To understand how this works, let's look at sampling and quantization more closely.

5.1.2.2 Sampling and Aliasing

Recall from Chapter 2 that the continuously changing air pressure of a single-frequency sound can be represented by a sine function, as shown in Figure 5.4. One cycle of the sine wave represents one cycle of compression and rarefaction of the sound wave. In digitization, a microphone detects changes in air pressure, sends corresponding voltage changes down a wire to an ADC, and the ADC regularly samples the values. The physical process of measuring the changing air pressure amplitude over time can be modeled by the mathematical process of evaluating a sine function at particular points across the horizontal axis.

Flash Tutorial: Sampling and Aliasing

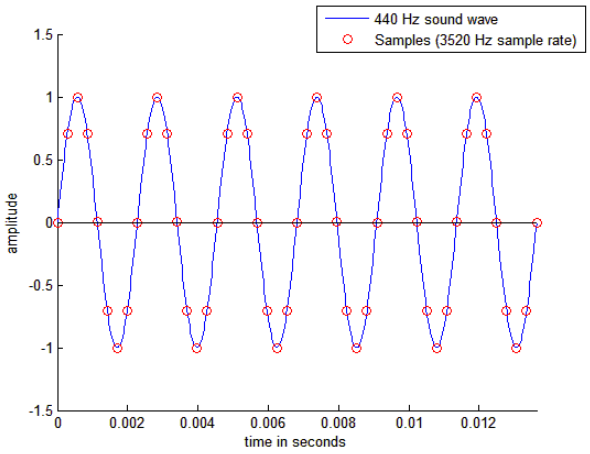


Figure 5.4 Graph of sine function modeling a 440 Hz sound wave

Figure 5.4 shows eight samples being taken for each cycle of the sound wave. The samples are represented as circles along the waveform. The sound wave has a frequency of 440 cycles/s (440 Hz), and the sampling rate has a frequency of 3520 samples/s (3520 Hz). The samples are stored as binary numbers. From these stored values, the amplitude of the digitized sound can be recreated and turned into analog voltage changes by the DAC.

The quantity of these stored values that exists within a given amount of time, as defined by the sampling rate, is important to capturing and recreating the frequency content of the audio signal. The higher the frequency content of the audio signal, the more samples per second (higher sampling rate) are needed to accurately represent it in the digital domain. Consider what would happen if only one sample was taken for every one-and-a-quarter cycles of the sound wave, as pictured in Figure 5.5. This would not be enough information for the DAC to correctly reconstruct the sound wave. Some cycles have been "jumped over" by the sampling process. In the figure, the higher-frequency wave is the original analog 440 Hz wave.

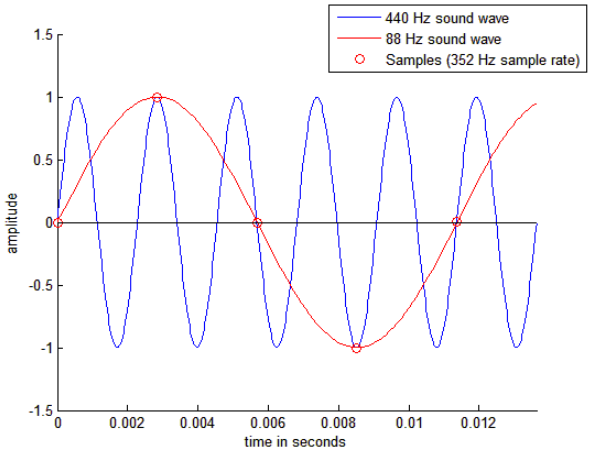


Figure 5.5 440 Hz wave undersampled

When the sampling rate is too low, the reconstructed sound wave appears to be lower-frequency than the original sound (or have an incorrect frequency component, in the case of a

complex sound wave). This is a phenomenon called **aliasing** – the incorrect digitization of a sound frequency component resulting from an insufficient sampling rate.

For a single-frequency sound wave to be correctly digitized, the sampling rate must be at least twice the frequency of the sound wave. More generally, for a sound with multiple frequency components, the sampling rate must be at least twice the frequency of the highest frequency component. This is known as the **Nyquist theorem**.

The Nyquist Theorem

Given a sound with maximum frequency component of *f* Hz, a sampling rate of at least *2f* is required to avoid aliasing. The minimum acceptable sampling rate (*2f* in this context) is called the **Nyquist rate**.

Given a sampling rate of *f,* the highest-frequency sound component that can be correctly sampled is *f/2*. The highest frequency component that can be correctly sampled is called the **Nyquist frequency**.

In practice, aliasing is generally not a problem. Standard sampling rates in digital audio recording environments are high enough to capture all frequencies in the human-audible range. The highest audible frequency is about 20,000 Hz. In fact, most people don't hear frequencies this high, as our ability to hear high frequencies diminishes with age. **CD quality sampling rate** is 44,100 Hz (44.1 kHz), which is acceptable as it is more than twice the highest audible component. In other words, with CD quality audio, the highest frequency we care about capturing (20 kHz for audibility purposes) is less than the Nyquist frequency for that sampling rate, so this is fine. A sampling rate of 48 kHz is also widely supported, and sampling rates go up as high as 192 kHz.

Even if a sound contains frequency components that are above the Nyquist frequency, to avoid aliasing the ADC generally filters them out before digitization.

Section 5.3.1 gives more detail about the mathematics of aliasing and an algorithm for determining the frequency of the aliased wave in cases where aliasing occurs.

5.1.2.3 Bit Depth and Quantization Error

When samples are taken, the amplitude at that moment in time must be converted to integers in binary representation. The number of bits used for each sample, called the **bit depth**, determines the precision with which you can represent the sample amplitudes. For this discussion, we assume that you know that basics of binary representation, but let's review briefly.

Binary representation, the fundamental language of computers, is a base 2 number system. Each *bit* in a binary number holds either a 1 or a 0. Eight bits together constitute one *byte*. The bit positions in a binary number are numbered from right to left starting at 0, as shown in Figure 5.6. The rightmost bit is called the **least significant bit**, and the leftmost is called the **most significant bit**. The i^{th} bit is called $b[i]$.

b[7]	b[6]	b[5]	b[4]	b[3]	b[2]	b[1]	b[0]
1	0	1	1	0	0	1	1
2^7=128	2^6=64	2^5=32	2^4=16	2^3=8	2^2=4	2^1=2	2^0=1

$$10110011_2 = 128 + 0 + 32 + 16 + 0 + 0 + 2 + 1 = 179_{10}$$

Figure 5.6 An 8-bit binary number converted to decimal

The value of an n-bit binary number is equal to

$$\sum_{i=0}^{n-1} b[i] * 2^i$$

Equation 5.1

Notice that doing the summation from $i = 0\ to\ n - 1$ causes the terms in the sum to be in the reverse order from that shown in Figure 5.6. The summation for our example is

$$10110011_2 = 1*1 + 1*2 + 0*4 + 0*8 + 1*16 + 1*32 + 0*64 + 1*128 = 179_{10}$$

Thus, 10110011 in base 2 is equal to 179 in base 10. (Base 10 is also called **decimal**.) We leave off the subscript 2 in binary numbers and the subscript 10 in decimal numbers when the base is clear from the context.

From the definition of binary numbers, it can be seen that the largest decimal number that can be represented with an n-bit binary number is $2^n - 1$, and the number of different values that can be represented is 2^n. For example, the decimal values that can be represented with an 8-bit binary number range from 0 to 255, so there are 256 different values.

These observations have significance with regard to the bit depth of a digital audio recording. A bit depth of 8 allows 256 different discrete levels at which samples can be recorded. A bit depth of 16 allows 2^{16} = 65,536 discrete levels, which in turn provides much higher precision than a bit depth of 8.

The process of quantization is illustrated Figure 5.7. Again, we model a single-frequency sound wave as a sine function, centering the sine wave on the horizontal axis. We use a bit depth of 3 to simplify the example, although this is far lower than any bit depth that would be used in practice. With a bit depth of 3, 2^3 = 8 quantization levels are possible. By convention, half of the quantization levels are below the horizontal axis (that is, 2^{n-1} of the quantization levels). One level is the horizontal axis itself (level 0), and $2^{n-1} - 1$ levels are above the horizontal axis. These levels are labeled in the figure, ranging from –4 to 3. When a sound is sampled, each sample must be scaled to one of 2^n discrete levels. However, the samples in reality might not fall neatly onto these levels. They have to be rounded up or down by some consistent convention. We round to the nearest integer, with the exception that values at 3.5 and above are rounded down to 3. The original sample values are represented by red dots on the graphs. The quantized values are represented as black dots. The difference between the original samples and the quantized samples constitutes rounding error. The lower the bit depth, the more values potentially must be rounded, resulting in greater quantization error. Figure 5.8 shows a simple view of the original wave vs. the quantized wave.

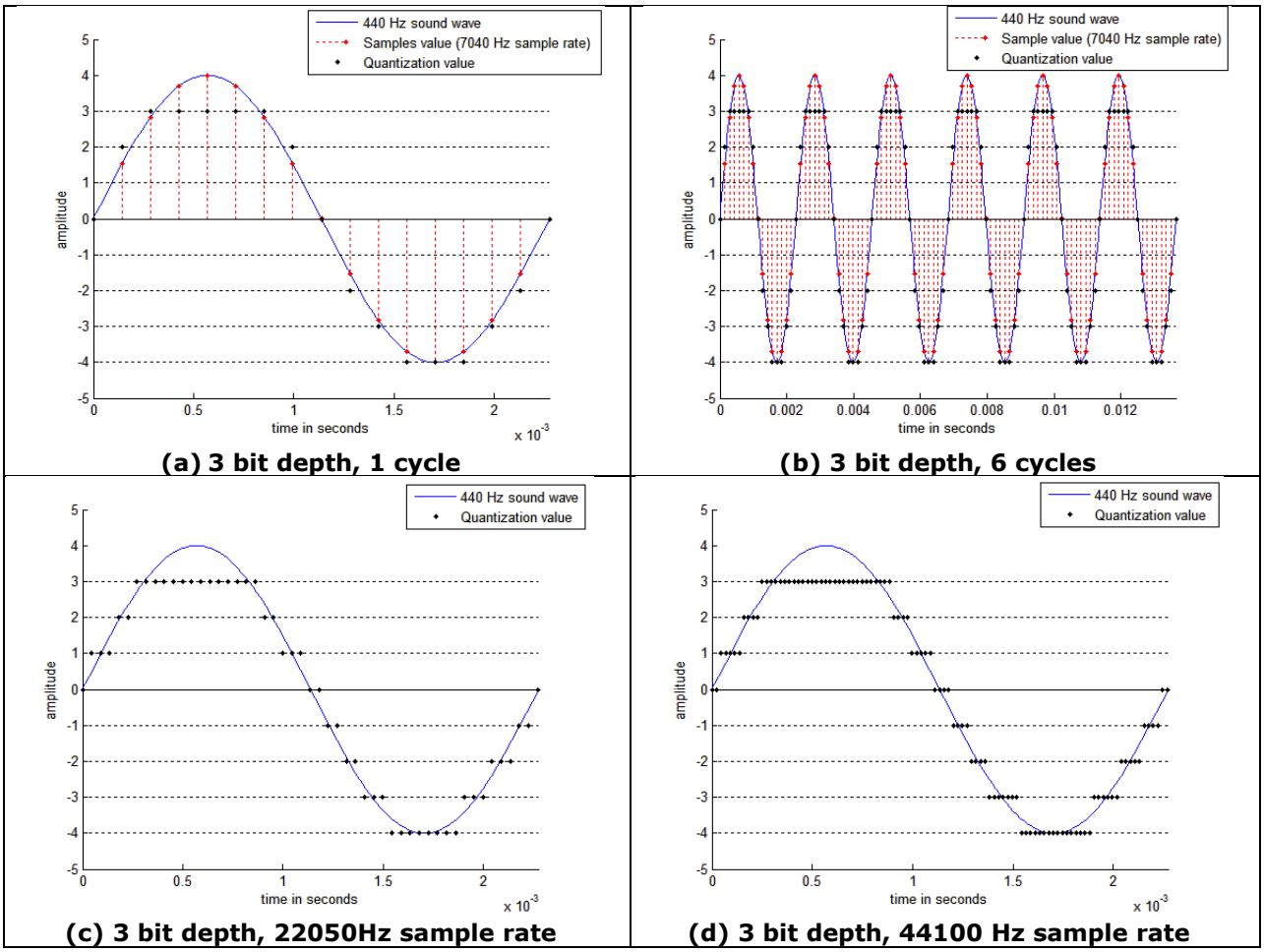


Figure 5.7 Quantization of a sampled sound wave

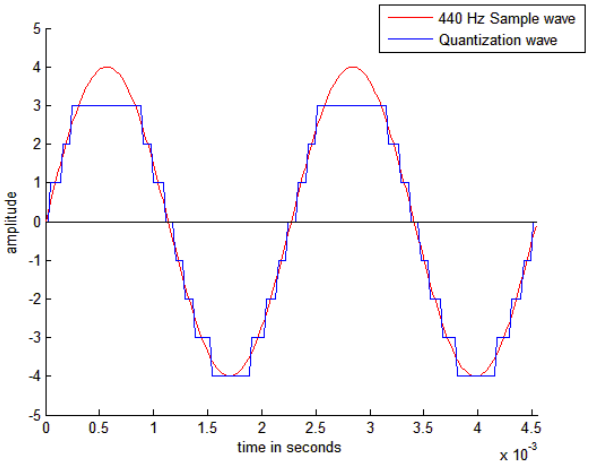


Figure 5.8 Quantized wave

Quantization error is sometimes referred to as noise. **Noise** can be broadly defined as part of an audio signal that isn't supposed to be there. However, some sources would argue that a better term for quantization error is **distortion**, defining distortion as an unwanted part of an

audio signal that is related to the true signal. If you subtract the stair-step wave from the true sine wave in Figure 5.8, you get the green part of the graphs in Figure 5.9. This is precisely the error – i.e., the distortion – resulting from quantization. Notice that the error follows a regular pattern that changes in tandem with the original "correct" sound wave. This makes the distortion sound more noticeable in human perception, as opposed to completely random noise. The error wave constitutes sound itself. If you take the sample values that create the error wave graph in Figure 5.9, you can actually play them as sound. You can compare and listen to the effects of various bit depths and the resulting quantization error in the Max Demo "Bit Depth" linked to this section.

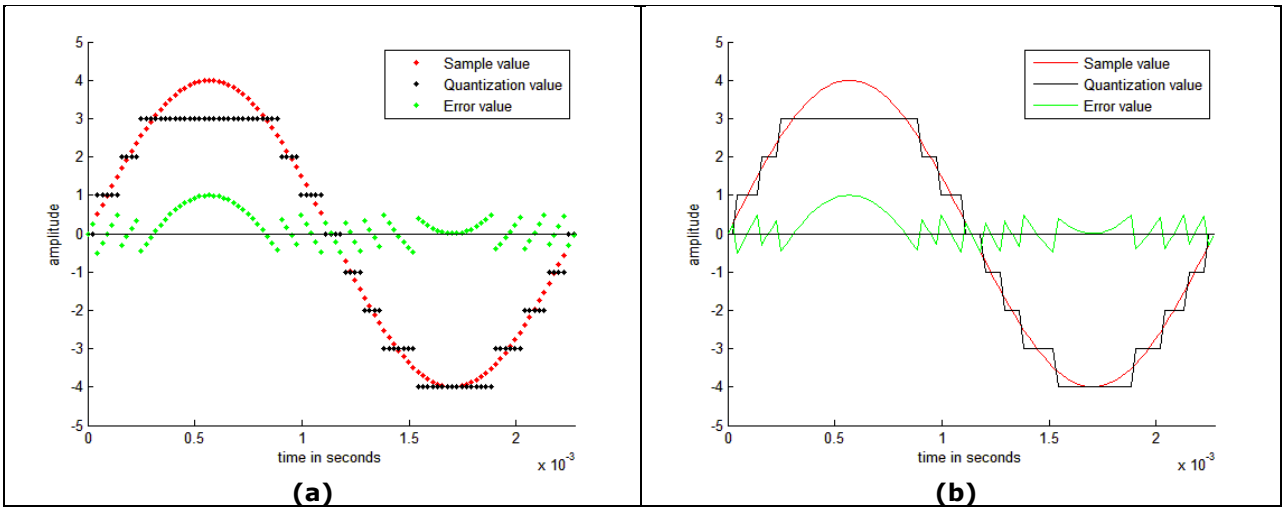


Figure 5.9 Error wave resulting from quantization

For those who prefer to distinguish between noise and distortion, noise is defined as an unwanted part of an audible signal arising from environmental interference – like background noise in a room where a recording is being made, or noise from the transmission of an audio signal along wires. This type of noise is more random than the distortion caused by quantization error. Section 3 shows you how you can experiment with sampling and quantization in MATLAB, C++, and Java programming to understand the concepts in more detail.

5.1.2.4 Dynamic Range

Another way to view the implications of bit depth and quantization error is in terms of dynamic range. The term **dynamic range** has two main usages with regard to sound. First, an occurrence of sound that takes place over time has a dynamic range, defined as the range between the highest and lowest amplitude moments of the sound. This is best illustrated by music. Classical symphonic music generally has a wide dynamic range. For example, "Beethoven's Fifth Symphony" begins with a high amplitude "Bump bump bump baaaaaaaa" and continues with a low-amplitude string section. The difference between the loud and quiet parts is intentionally dramatic and is what gives the piece a wide dynamic range. You can see this in the short clip of the symphony graphed in Figure 5.10. The dynamic range of this clip is the difference between the magnitude of the largest sample value and the magnitude of the smallest. Notice that you don't measure the range *across* the horizontal access but from the highest-magnitude sample either above or below the axis to the lowest-magnitude sample on the same side of the axis. A

sound clip with a narrow dynamic range has a much smaller difference between the loud and quiet parts. In this usage of the term dynamic range, we're focusing on the **dynamic range of a particular occurrence of sound or music**.

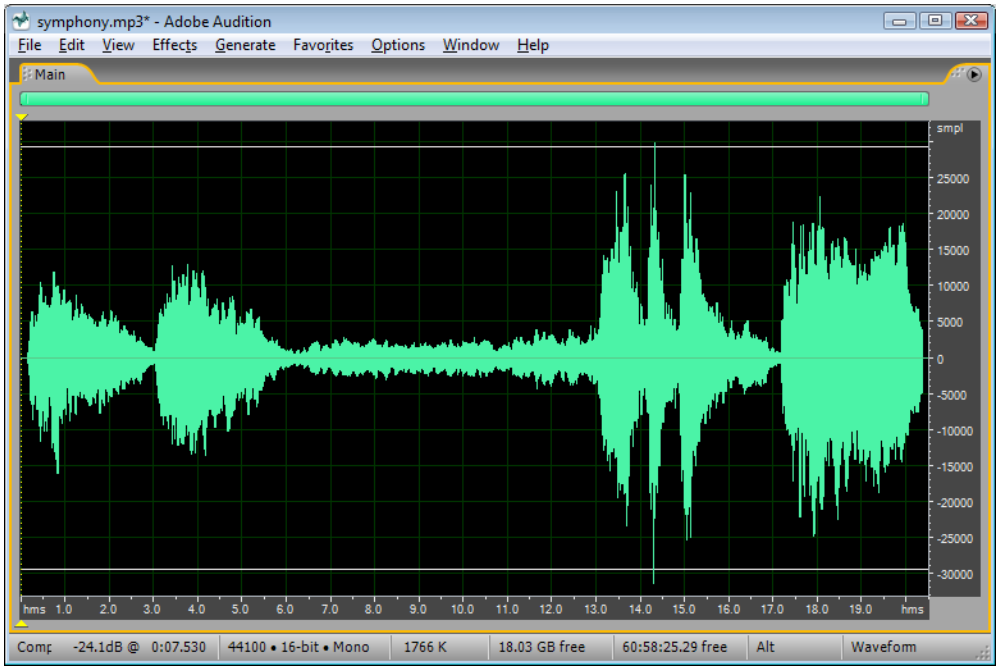


Figure 5.10 Music with a wide dynamic range

In another usage of the term, the **potential dynamic range of a digital recording** refers to the possible range of high and low amplitude samples as a function of the bit depth. Choosing the bit depth for a digital recording automatically constrains the dynamic range, a higher bit depth allowing for a wider dynamic range.

Consider how this works. A digital recording environment has a maximum amplitude level that it can record. On the scale of *n*-bit samples, the maximum amplitude (in magnitude) would be 2^{n-1}. The question is this: How far,

Aside:
MATLAB Code for Figure 5.11 and Figure 5.12:

```
hold on;
f = 440;T = 1/f;
Bdepth = 3; bit depth
Drange = 2^(Bdepth-1); dynamic range
axis = [0 2*T -(Drange+1) Drange+1];
SRate = 44100; %sample rate
sample_x = (0:2*T*SRate)./SRate;
sample_y = Drange*sin(2*pi*f*sample_x);
plot(sample_x,sample_y,'r-');
q_y = round(sample_y); %quantization value
for i = 1:length(q_y)
    if q_y(i) == Drange
       q_y(i) = Drange-1;
    end
end
plot(sample_x,q_y,'-')
for i = -Drange:Drange-1 %quantization level
    y = num2str(i); fplot(y,axis,'k:')
end
legend('Sample wave','Quantization
wave','Quantization level')
y = '0'; fplot(y,axis,'k-')
ylabel('amplitude');xlabel('time in seconds');
hold off;
```

relatively, can the audio signal fall below the maximum level before it is rounded down to silence? The figures below show the dynamic range at a bit depth of 3 (Figure 5.11) compared to the dynamic range at a bit depth of 4 (Figure 5.12). Again, these are non-practical bit depths chosen for simplicity of illustration. The higher bit depth gives a wider range of sound amplitudes that can be recorded. The smaller bit depth loses more of the quiet sounds when they are rounded down to zero or overpowered by the quantization noise. You can see this in the right hand view of Figure 5.13, where a portion of "Beethoven's Ninth Symphony" has been reduced to four bits per sample.

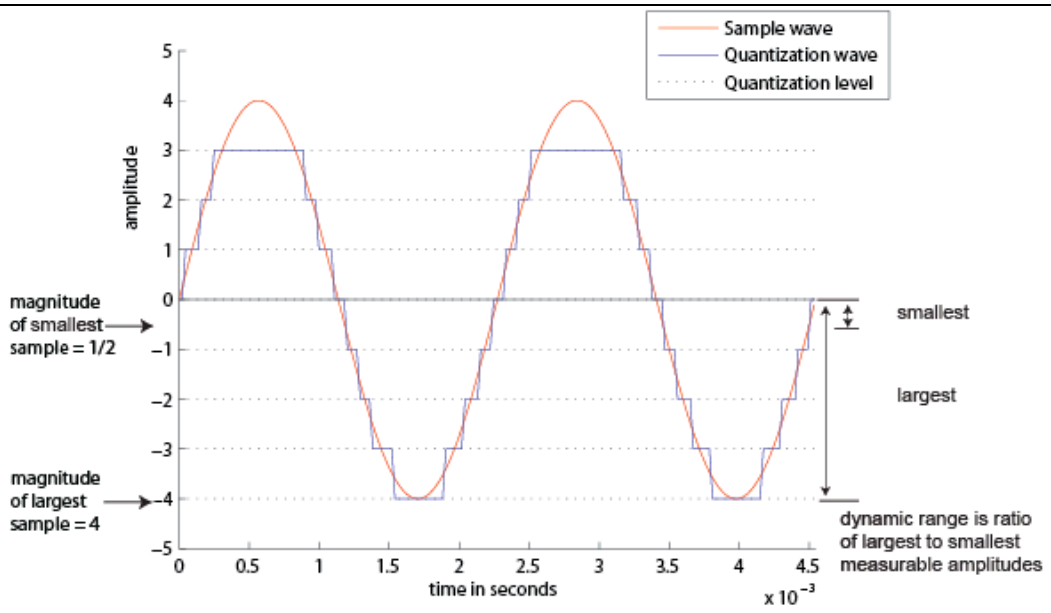


Figure 5.11 Dynamic range for wave quantized at 3 bits

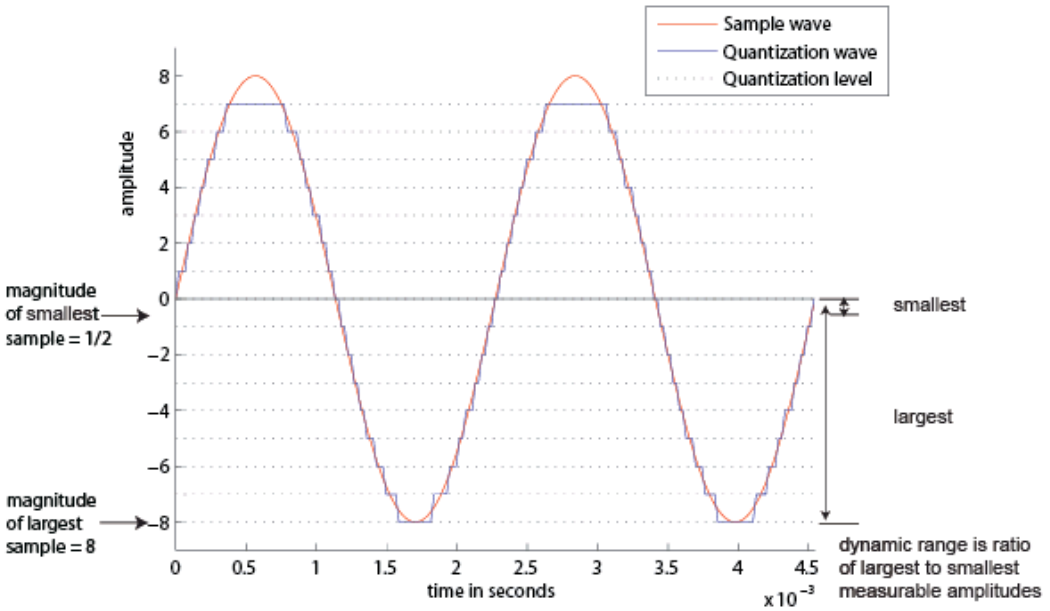
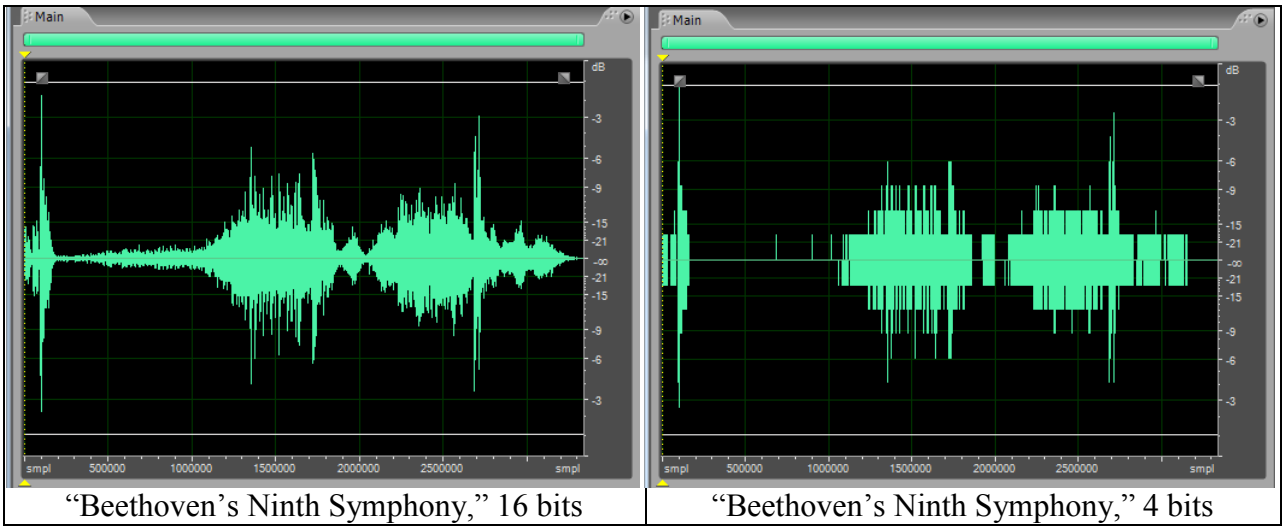


Figure 5.12 Dynamic range for wave quantized at 4 bits



"Beethoven's Ninth Symphony," 16 bits | "Beethoven's Ninth Symphony," 4 bits

Figure 5.13 Sample values falling to 0 due to insufficient bit depth

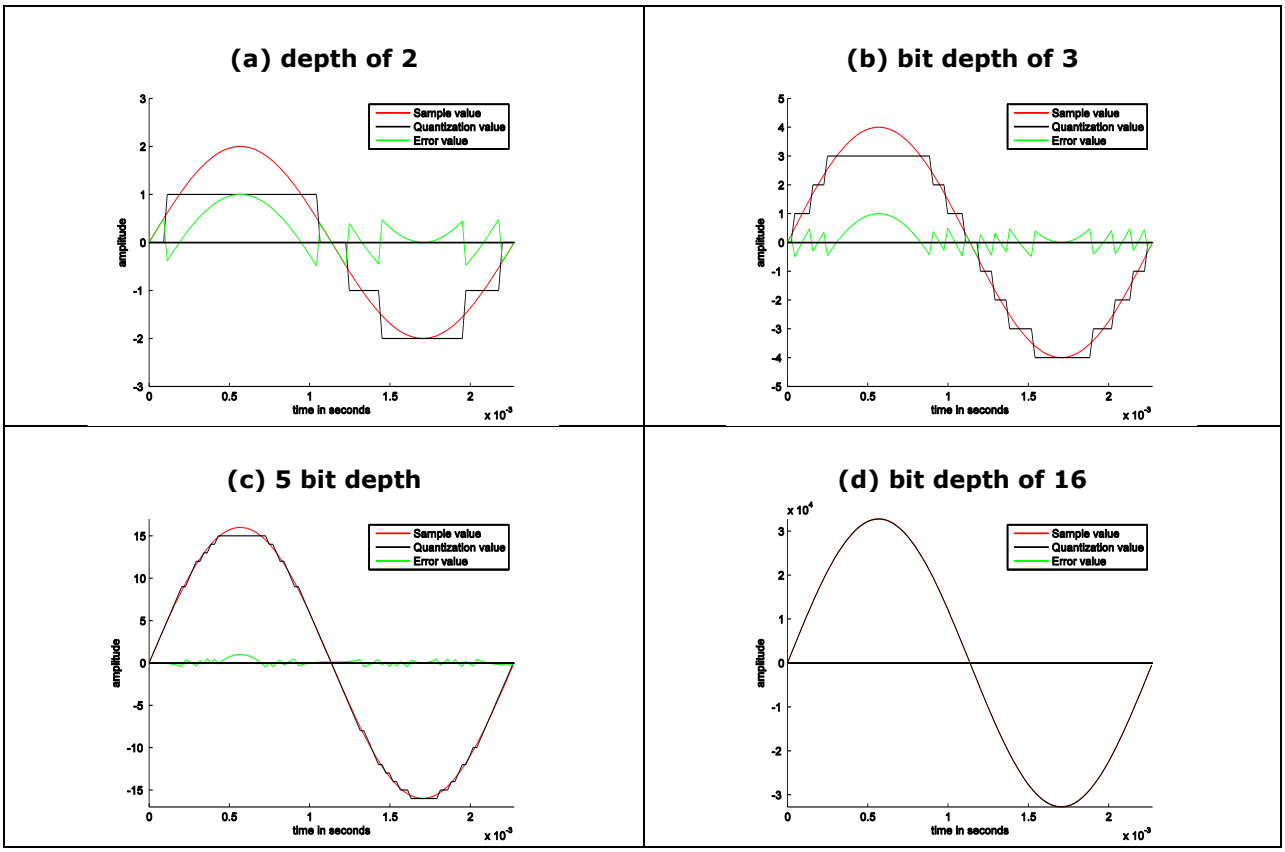


Figure 5.14 Wave quantized at different bit depths

The term *dynamic range* has a precise mathematical definition that relates back to quantization error. In the context of music, we encountered dynamic range as the difference between the loudest and quietist musical passages of a song or performance. Digitally, the loudest potential passage of the music (or other audio signal) that could be represented would

have full amplitude (all bits on). The quietest possible passage is the quantization noise itself. Any sound quieter than that would simply be masked by the noise or rounded down to silence. The difference between the loudest and quietest parts is therefore the highest (possible) amplitude of the audio signal compared to the amplitude of the quantization noise, or the ratio between the signal level to the noise level. This is what is known as signal-to-quantization-noise-ratio (SQNR), and in this context, dynamic range is the same thing. This definition is given in Equation 5.2.

Max Demo: Bit Depth

Given a bit depth of n, the dynamic range of a digital audio recording is equal to $20log_{10}\left(\frac{2^{n-1}}{1/2}\right)$ dB.

Equation 5.2

You can see from the equation that dynamic range as SQNR is measured in decibels. Decibels are a dimensionless unit derived from the logarithm of the ratio between two values. For sound, decibels are based on the ratio between the air pressure amplitude of a given sound and the air pressure amplitude of the threshold of hearing. For dynamic range, decibels are derived from the ratio between the maximum and minimum amplitudes of an analog waveform quantized with n bits. When values are quantized by rounding down to the nearest integer, the maximum magnitude amplitude is $|-2^{n-1}| = -2^{n-1} - 1$. The minimum amplitude of an analog wave that would be converted to a non-zero value when it is quantized is ½. Signal-to-quantization-noise is based on the ratio between these maximum and minimum values for a given bit depth. It turns out that this is exactly the same value as the dynamic range.

Equation 5.2 can be simplified as shown in Equation 5.3.

$$20log_{10}\left(\frac{2^{n-1}}{1/2}\right) = 20log_{10}(2^n) = 20nlog_{10}(2) \approx 6.04n$$

Equation 5.3

Flash Tutorial: Dynamic Range

Equation 5.3 gives us a method for determining the possible dynamic range of a digital recording as a function of the bit depth. For bit depth n, the possible dynamic range is approximately $6n$ dB. A bit depth of 8 gives a dynamic range of approximately 48 dB, a bit depth of 16 gives a dynamic range of about 96 dB, and so forth.

When we introduced this section, we added the adjective "potential" to "dynamic range" to emphasize that it is the *maximum possible range* that can be used as a function of the bit depth. But not all of this dynamic range is used if the amplitude of a digital recording is relatively low, never reaching its maximum. Thus, we it is important to consider the **actual dynamic range (or actual SQNR)** as opposed to the **potential dynamic range (or potential SQNR**, just defined). Take, for example, the situation illustrated in Figure 5.15. A bit depth of 7 has been chosen. The amplitude of the wave is 24 dB below the maximum possible. Because the sound uses so little of its potential dynamic range, the actual dynamic range is small. We've used just a simple sine wave in this example so that you can easily see the error wave in proportion to the sine wave, but you can imagine a music recording that has a low actual dynamic range because the recording was done at a low level of amplitude.

The difference between potential dynamic range and actual dynamic range is illustrated in detail in the interactive Max demo associated with this section. This demo shows that, in addition to choosing a bit depth appropriately to provide sufficient dynamic range for a sound recording, it's important that you use the available dynamic range optimally. This entails setting microphone input voltage levels so that the loudest sound produced is as close as possible to the maximum recordable level. These practical considerations are discussed further in Section 5.2.2.

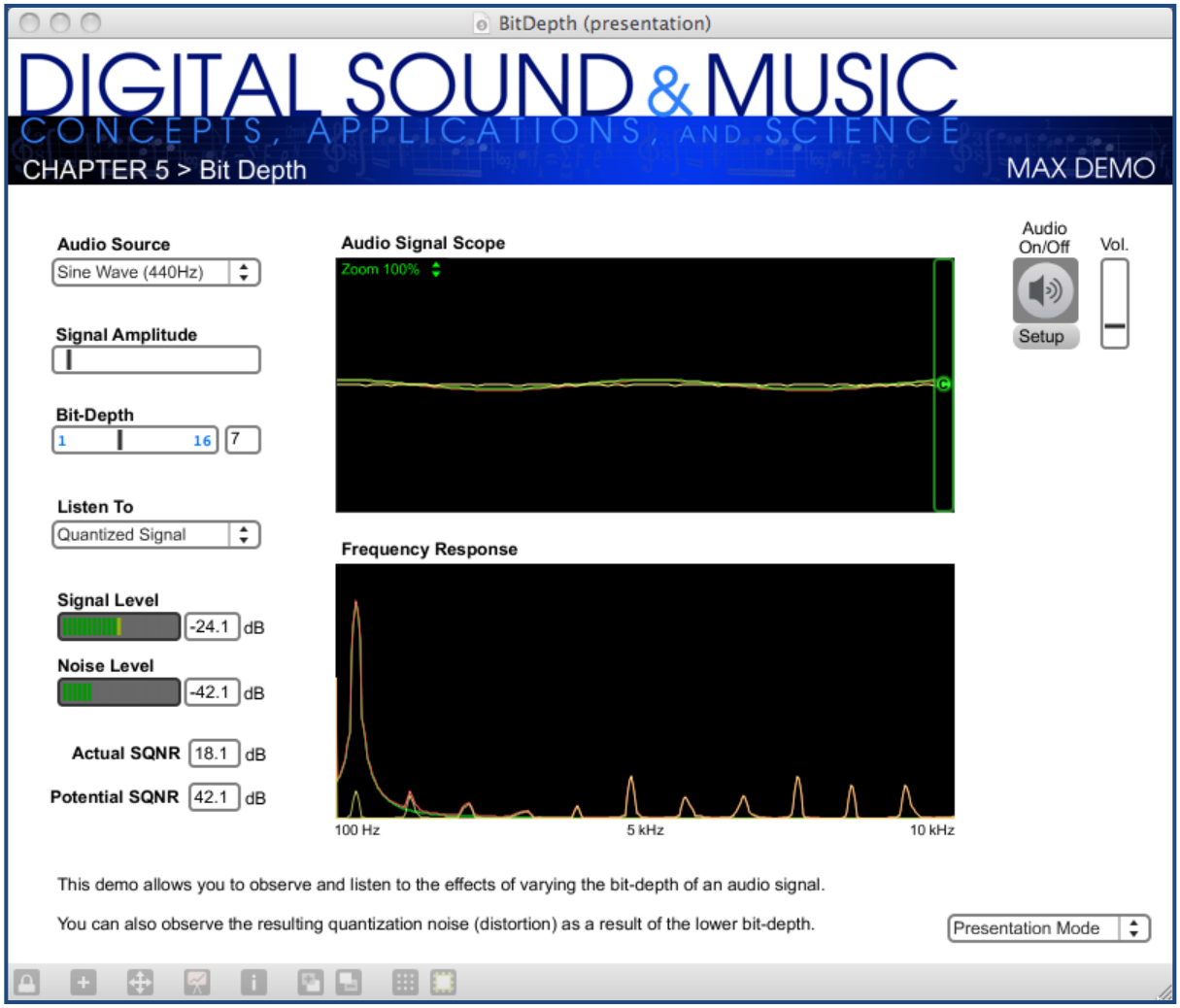


Figure 5.15 Potential SQNR compared to actual SQNR

We have one more related usage of decibels to define in this section. In the interface of many software digital audio recording environments, you'll find that **decibels-full-scale** (**dBFS**) is used. (In fact, it is used in the Signal and Noise Level meters in Figure 5.15.) As you can see in Figure 5.16, which shows amplitude in dBFS, the maximum amplitude is 0 dBFS, at equidistant positions above and below the horizontal axis. As you move toward the horizontal axis (from either above or below) through decreasing amplitudes, the dBFS values become increasingly negative.

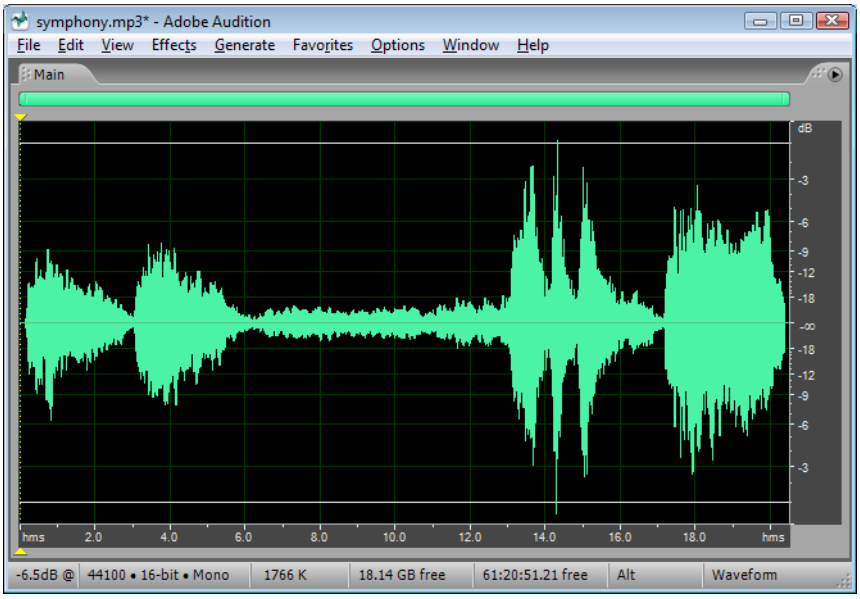


Figure 5.16 Dynamic range shown in dBFS

The equation for converting between dB and dBFS is given in Equation 5.4

For n-bit samples, **decibels-full-scale** (**dBFS**) is defined as follows:

$$dBFS = 20log_{10}\left(\frac{x}{2^{n-1}}\right)$$

where x is an integer sample value between 0 and 2^{n-1}.

Equation 5.4

Generally, computer-based sample editors allow you to select how you want the vertical axis labeled, with choices including sample values, percentage, values normalized between –1 and 1, and dBFS.

Chapter 7 goes into more depth about dynamics processing, the adjustment of dynamic range of an already-digitized sound clip.

5.1.2.5 Audio Dithering and Noise Shaping

It's possible to take an already-recorded audio file and reduce its bit depth. In fact, this is commonly done. Many audio engineers keep their audio files at 24 bits while they're working on them, and reduce the bit depth to 16 bits when they're finished processing the files or ready to burn to an audio CD. The advantage of this method is that when the bit depth is higher, less error is introduced by processing steps like normalization or adjustment of dynamics. Because of this advantage, even if you choose a bit depth of 16 from the start, your audio processing system may be using 24 bits (or an even higher bit depth) behind the scenes anyway during processing, as is the case with Pro Tools.

Audio dithering is a method to reduce the quantization error introduced by a low bit depth. Audio dithering can be used by an ADC when quantization is initially done, or it can be used on an already-quantized audio file when bit depth is being reduced. Oddly enough,

dithering works by adding a small amount of random noise to each sample. You can understand the advantage of doing this if you consider a situation where a number of consecutive samples would all round down to 0 (i.e., silence), causing breaks in the sound. If a small random amount is added to each of these samples, some round up instead of down, smoothing over those breaks. In general, dithering reduces the perceptibility of the distortion because it causes the distortion to no longer follow exactly in tandem with the pattern of the true signal. In this situation, low-amplitude noise is a good trade-off for distortion.

Noise shaping is a method that can be used in conjunction with audio dithering to further compensate for bit-depth reduction. It works by raising the frequency range of the rounding error after dithering, putting it into a range where human hearing is less sensitive. When you reduce the bit depth of an audio file in an audio processing environment, you are often given an option of applying dithering and noise shaping, as shown in Figure 5.17 Dithering can be done without noise shaping, but noise shaping is applied only after dithering. Also note that dithering and noise shaping cannot be done apart from the requantization step because they are embedded into the way the requantization is done. A popular algorithm for dithering and noise shaping is the proprietary POW-r (Psychoacoustically Optimized Wordlength Reduction), which is built into Pro Tools, Logic Pro, Sonar, and Ableton Live.

Dithering and noise shaping are discussed in more detail in Section 5.3.7.

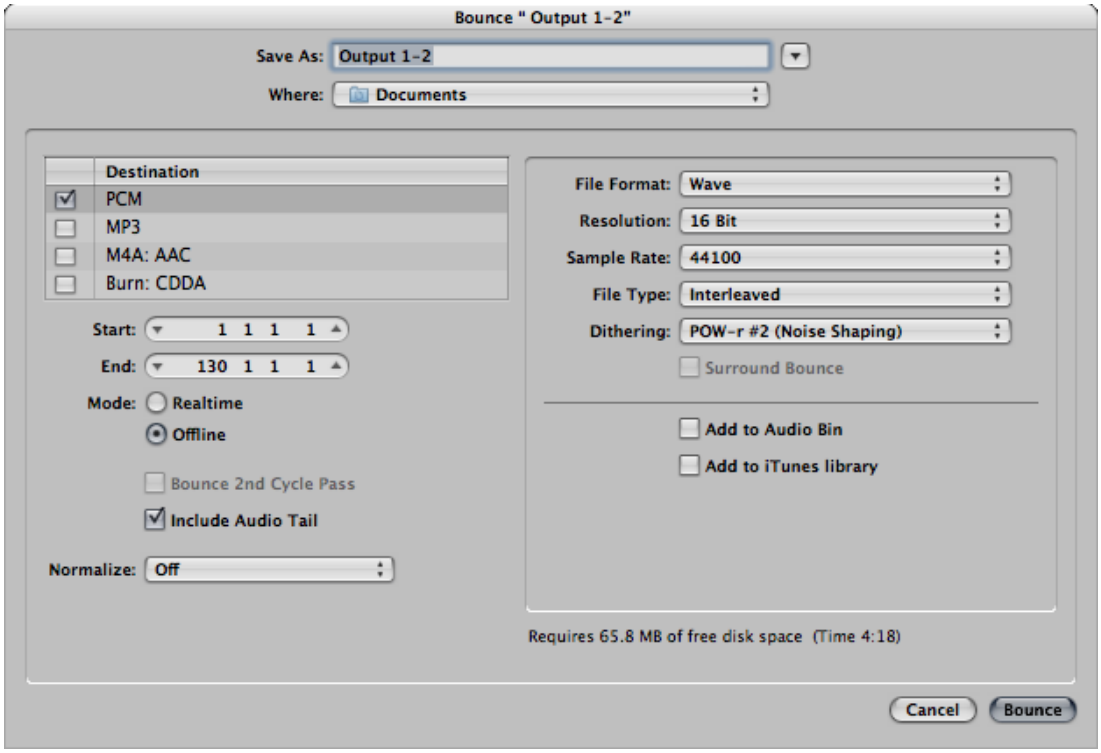


Figure 5.17 Changing bit depth with dither in Logic

5.1.3 Audio Data Streams and Transmission Protocols

Audio data passed from one device to another is referred to as a **stream**. There are several different formats for transmitting a digital audio stream between devices. In some cases, you might want to interconnect two pieces of equipment digitally, but they don't offer a compatible transmission protocol. The best thing to do is make sure you purchase equipment that is

compatible with the equipment you already have. In order to succeed at this, you'll need to understand and be able to identify the various options for digital audio transmission.

The most common transmission protocol, particularly in consumer-grade equipment is the Sony/Phillips Digital Interconnect Format (S/PDIF). With **S/PDIF** you can transmit two channels on a single wire. Typically this means you can transmit both the left and right channels of a stereo pair using a single cable instead of two cables required for analog transmission. S/PDIF can be transmitted electrically or optically. Electrical transmission involves **RCA** (Radio Corporation of America) connectors and a low loss, high bandwidth coaxial cable. This cable is different from the cable you would use for analog transmission. For S/PDIF you need a cable like what is used for video. S/PDIF transmits the digital data electrically in a stream of square wave pulses. Using cheap, low bandwidth cable can result in a loss of high frequency content that can ultimately lose the square wave form, resulting in data loss. When looking for a cable for electrical S/PDIF transmission, look for something with RCA connectors on each end and an impedance of 75 Ω.

S/PDIF can also be transmitted optically using an optical cable with **TOSLINK** (TOShiba-LINK) connectors. Optical transmission has the advantage of being able to run longer distances without the risk of signal loss, and it is not susceptible to electromagnetic interference like an electrical signal. Optical cables are more easily broken so if you plan to move your equipment around, you should invest in an optical cable that has good insulation.

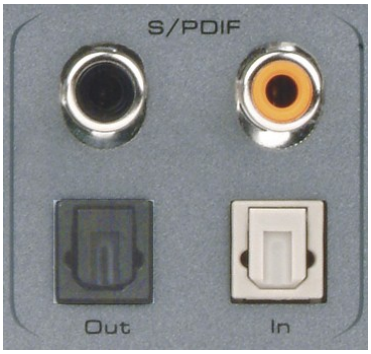


Figure 5.18 - S/PDIF connections using RCA and TOSLINK connections

S/PDIF is considered a consumer grade transmission protocol. S/PDIF has a professional grade cousin called **AES3**, more commonly known as **AES/EBU** (Audio Engineering Society/European Broadcasting Union). The actual format of the digital stream is almost identical. The main differences are the type of cable and connectors used and the maximum distance you can reliably transmit the signal. AES/EBU can be run electrically as a balanced signal using three pin XLR connectors with a 110 Ω twisted pair cable or unbalanced using BNC connectors a with a 75 Ω coaxial cable. The unbalanced version has a maximum transmission distance of 1000 meters as opposed to the 100 meters maximum for the balanced version. The balanced signal is by far the most common implementation as shown in Figure 5.19.

Figure 5.19 - AES/EBU connections using XLR connectors

If you need to transmit more than two channels of digital audio between devices, there are several options available. The most common is the **ADAT Optical Interface** (Alesis Digital Audio Tape). Alesis developed the system to allow signal transfer between their eight-track digital audio tape recorders, but the system has since been widely adopted for multi-channel digital signal transmission between devices at short distances. ADAT can transmit eight channels of audio at sampling rates up to 48 kHz or four channels at sampling rates up to 96 kHz. ADAT uses the same optical TOSLINK cable used for S/PDIF. This makes it relatively inexpensive for the consumer. However, the protocol must be licensed from Alesis if a manufacturer wants to implement it in their equipment.

There are several other emerging standards for multi-channel digital audio transmission, more than we can cover in the scope of this chapter. What is important to know is that most protocols allow digital transmission of 64 channels or more of digital audio over long distances using fiber optic, or CAT-5e cable. Examples of this kind of transmission include MADI, AVB, CobraNet, A-Net, and mLAN. If you need this level of functionality, you will likely be able to purchase interface cards that use these protocols for most computers and digital mixing consoles.

5.1.4 Signal Path in an Audio Recording System

In Section 5.1.1, we illustrated a typical setup for a computer-based digital audio recording and editing system. Let's look at this more closely now, with special attention to the signal path and conversions of the audio stream between analog and digital forms.

Figure 5.20 illustrates a recording session where a singer is singing into a microphone and monitoring the recording session with headphones. As the audio stream makes its way along the signal path, it passes through a variety of hardware and software, including the microphone, audio interface, audio driver, CPU, input and output buffers (which could be hardware or software), and hard drive. The **CPU** (**central processing unit**) is the main hardware workhorse of a computer, doing the actual computation that is required for tasks like accepting audio streams, running application programs, sending data to the hard drive, sending files to the printer, and so forth. The CPU works hand-in-hand with the **operating system**, which is the software program that manages which task is currently being worked on, like a conductor conducting an orchestra. Under the direction of the operating system, the CPU can give little slots of times to various processes, swapping among these processes very quickly so that it looks like each one is advancing normally. This way, multiple processes can appear to be running simultaneously.

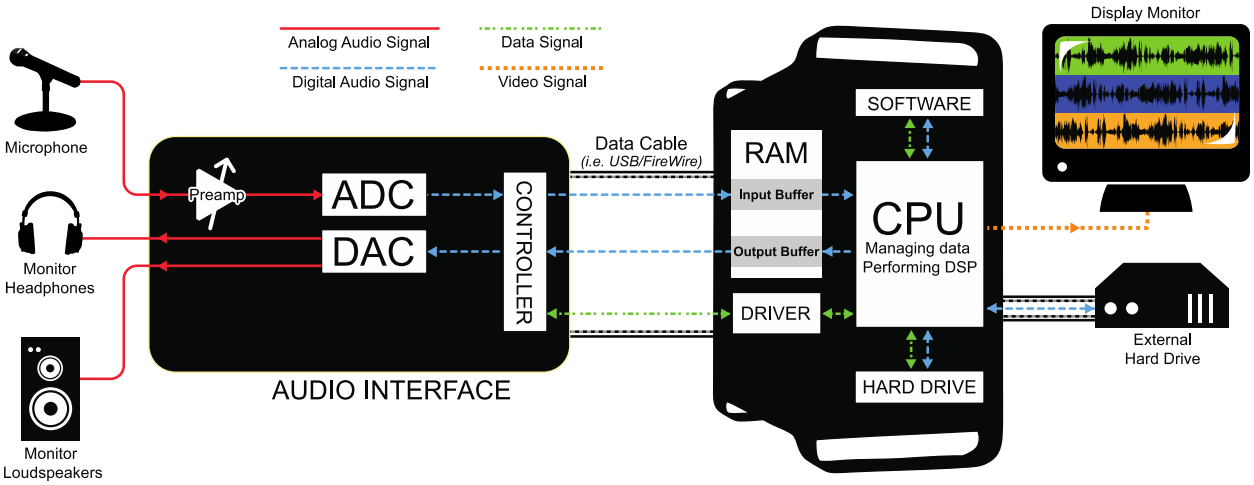


Figure 5.20 Signal path in digital audio recording

Now let's get back to how a recording session works. During the recording session, the microphone, an analog device, receives the audio signal and sends it to the audio interface in analog form. The audio interface could be an external device or an internal sound card. The audio interface performs analog-to-digital conversion and passes the digital signal to the computer. The audio signal is received by the computer in a digital stream that is interpreted by a driver, a piece of software that allows two hardware devices to communicate. When an audio interface is connected to a computer, an appropriate driver must be installed in the computer so that the digital audio stream generated by the audio interface can be understood and located by the computer.

The driver interprets the audio stream and sends it to an audio input buffer in RAM. Saying that the audio buffer is in RAM implies that this is a software buffer. (The audio interface has a small hardware buffer as well, but we don't need to go to this level of detail.) The **audio input buffer** provides a place where the audio data can be held until the CPU is ready to process it. Buffering of audio input is necessary because the CPU may not be able to process the audio stream as soon as it comes into the computer. The CPU has to handle other processes at the same time (monitor displays, operating system tasks, etc.). It also has to make a copy of the audio data on the hard disk because the digitizing and recording process generates too much data to fit in RAM. Moving back and forth to the hard drive is time consuming. The input buffer provides a holding place for the data until the CPU is ready for it.

Let's follow the audio signal now to its output. In Figure 5.20, we're assuming that you're working on the audio in a program like Sonar or Logic. These audio processing programs serve as the user interface for the recording process. Here you can specify effects to be applied to tracks, start and stop the recording, and decide when and how to save the recording in permanent storage. The CPU performs any digital signal processing (DSP) called for by the audio processing software. For example, you might want reverb added to the singer's voice. The CPU applies the reverb and then sends the processed data to the **software output buffer**. From here the audio data go back to the audio interface where it is converted back to analog format and sent to the singer's headphones or a set of connected loudspeakers. Possibly, the track of the singer's voice could be mixed with a previously recorded instrumental track before it is sent to the audio interface and then to the headphones.

For this recording process to run smoothly -- without delays between when you speak into the microphone and without breaks in the audio – you must choose and configure your

drivers and hard drives appropriately. These practical issues are discussed further in Section 5.2.3.

Figure 5.20 gives an overview of the audio signal path in one scenario, but there are many details and variations that have not been discussed here. We're assuming a software system such as Logic or Sonar is providing the mixer and DSP, but it's possible for additional hardware to play these roles – a hardware mixing console, equalizer, or dynamics compressor, for example. Some professional grade software applications may even have additional external DSP processors to help offload some of the work from the CPU itself, such as with Pro Tools HD. This external gear could be analog or digital, so additional DAC/ADC conversions might be necessary. We also haven't considered details like microphone pre-amps, loudspeakers and loudspeaker amps, data buses, and multiple channels. We refer the reader to the references at the end of the chapter for more information.

Flash Tutorial: Audio Signal Path

5.1.5 CPU and Hard Drive Considerations

As you work with digital audio, it's important that you have some understanding of the demands on your computer's CPU and hard drive.

In general, the CPU is responsible for running all active processes on your computer. Processes take turns getting slices of time from the CPU so that they can all be making progress in their execution at the same time. You might have processes running on your computer at the same time you're recording and not even be aware of it. For example, you may have automatic software updates activated. What if an automatic update tries to run while audio capture is in progress? The CPU is going to have to take some time to deal with that. If it's dealing with a software update, it isn't dealing with your audio stream.

Even if you make your best effort to turn off all other processes during recording, the CPU still has other important work to do that keeps it from returning immediately to the audio buffer. One of the CPU's most important responsibilities during audio recording is writing audio data out to the hard drive. To understand how this works, it may be helpful to review briefly the different roles of RAM and hard disk memory with regard to a running program.

Ordinarily, we think of software programs operating according to the following scenario. Data is loaded into RAM, which is a space in memory *temporarily* allocated to a certain program. The program does computation on the data, changing it in some way. Since the RAM allocated to the program is released when the program finishes its computation, the data must be written out to the hard disk if you want a permanent copy of it. In this sense, RAM is considered volatile memory. For all intents and purposes, the data in RAM disappears when the program finishes execution.

But what if the program you're running is an audio processing program like Logic or Sonar through which you're recording audio? The recording process causes audio data to be read into RAM. Why can't Sonar or Logic just store all the audio data in RAM until you're done working on it and write it out to the hard disk when you've finished? The problem is that a very large amount of data is generated as the recording is being done – 176,400 bytes per second for CD quality audio. For a recording of any significant length, it isn't feasible to store all the audio data in RAM. Thus, audio samples are constantly pulled from the RAM buffer by the CPU and placed on a hard drive. This takes a significant amount of the CPU's time.

In order to keep up efficiently with the constant stream of data, the hard drive needs a dedicated space in which to write the audio data. Most audio recording programs automatically claim a large portion of hard drive space when you start recording. The size of this hard drive allocation is usually controllable in the preferences of your audio software. For example, if your software is configured to allocate 500 MB of hard drive space for each audio stream, 500 MB is immediately claimed on the hard drive when you start recording, and no other software may write to that space. If your recording uses 100 MB, the operating system returns the remaining 400 MB of space to be available to other programs. It's important to make sure you configure the software to claim an appropriate amount of space. If your recording ends up needing more than 500 MB, the software begins dynamically finding new chunks of space on the hard drive to put the extra data. In this situation, it's possible for dropouts to happen if sufficient space cannot be found fast enough. The problem is compounded by multitrack recording. Imagine what happens if you're recording 24 tracks of audio at one time. The software has to find 24 blocks of space on the hard drive that are 500 MB in size. That's 12 GB of free space that needs to be immediately and exclusively available.

One way to avoid problems is to dedicate a separate hard drive other than your startup drive for audio capture and data storage. That way you know that no other program will attempt to use the space on that drive. You also need to have a hard drive that is large enough and fast enough to handle this amount of sustained data being written, and often read back. In today's technology, at least a 7200 RPM, one terabyte dedicated external hard drive with a high-speed connection should be sufficient.

5.1.6 Digital Audio File Types

You saw in the previous section that the digital audio stream moves through various pieces of software and hardware during a recording session, but eventually you're going to want to save the stream as a file in permanent storage. At this point you have to decide the format in which to save the file. Your choice depends on how and where you're going to use the recording.

> **Aside:** In our discussion of file types, we'll use capital letters like AIFF and WAV to refer to different formats. Generally there is a corresponding suffix, called a **file extension**, used on file names – e.g., *.aiff* or *.wav*. However, in some cases, more than one suffix can refer to the same basic file type. For example, *.aiff* and *.aif*, and *.aifc* are all variants of the AIFF format.

Audio file formats differ along a number of axes. They can be free or proprietary, platform-restricted or cross-platform, compressed or uncompressed, container files or simple audio files, and copy-protected or unprotected. (Copy protection is more commonly referred to as **digital rights management** or **DRM**.)

Proprietary file formats are controlled by a company or an organization. The particulars of a proprietary format and how the format is produced are not made public and their use is subject to patents. Some proprietary files formats are associated with commercial software for audio processing. Such files can be opened and used only in the software with which they're associated. Some examples are CWP files for Cakewalk Sonar, SES for Adobe Audition multitrack sessions, AUP projects for Audacity, and PTF for Pro Tools. These are project file formats that include meta-information about the overall organization of an audio project. Other proprietary formats – e.g., MP3 – may have patent restrictions on their use, but they have openly

documented standards and can be licensed for use on a variety of platforms. As an alternative, there exist some free, open source audio file formats, including OGG and FLAC.

Platform-restricted files can be used only under certain operating systems. For example, WMA files run under Windows, AIFF files run under Apple OS, and AU files run under Unix and Linux. The MP3 format is cross-platform. AAC is a cross-platform format that has become widely popular from its use on phones, pad computers, digital radio, and video game consoles.

The basic format for uncompressed audio data is called **PCM** (**pulse code modulation**). The term *pulse code modulation* is derived from the way in which raw audio data is generated and communicated. That is, it is generated by the process of sampling and quantization described in Section 5.1 and communicated as binary data by electronic pulses representing 0s and 1s. WAV, AIFF, AU, RAW, and PCM files can store uncompressed audio data. RAW files contain only the audio data, without even a header on the file.

> **Aside:** Pulse code modulation was introduced by British scientist A. Reeves in the 1930s. Reeves patented PCM as a way of transmitting messages in "amplitude-dichotomized, time-quantized" form – what we now call "digital."

You can't tell from the file extension whether or not a file is compressed, and if it is compressed, you can't necessarily tell what compression algorithm (called a **codec**) was used. There are both compressed and uncompressed versions of WAV, AIFF, and AU files. When you save an audio file, you can choose which type you want.

One basic reason that WAV and AIFF files come in compressed and uncompressed versions is that, in reality, these are **container file formats** rather than simple audio files. A container file wraps a simple audio file in a meta-format which specifies blocks of information that should be included in the header along with the size and position of chunks of data following the header. The container file may allow options for the format of the actual audio data, including whether or not it is compressed. If the audio is compressed, the system that tries to open and play the container file must have the appropriate codec in order to decompress and play it. AIFF files are container files based on a standardized format called IFF. WAV files are based on the RIFF format. MP3 is a container format that is part of the more general MPEG standard for audio and video. WMA is a Windows container format. OGG is an open source, cross-platform alternative.

In addition to audio data, container files can include information like the names of songs, artists, song genres, album names, copyrights, and other annotations. The metadata may itself be in a standardized format. For example, MP3 files use the **ID3** format for metadata.

Compression is inherent in some container file types and optional in others. MP3, AAC, and WMA files are always compressed. Compression is important if one of your main considerations is the ability to store lots of files. Consider the size of a CD quality audio file, which consists of two channels of 44,100 samples per second with two bytes per sample. This gives

$$2 * \frac{44100\ samples}{sec} * \frac{2\ bytes}{sample} * 60\ \frac{sec}{min} * 5\ \text{min} = 52{,}920{,}000\ bytes\ \approx\ 50.5\ MB$$

A five minute piece of music, uncompressed, takes up over 50 MB of memory. MP3 and AAC compression can reduce this to less than a tenth of the original size. Thus, MP3 files are popular for portable music players, since compression makes it possible to store many more songs.

Compression algorithms are of two basic types: lossless or lossy. In the case of a **lossless compression algorithm**, no audio information is lost from compression to decompression. The audio information is compressed, making the file smaller for storage and transmission. When it is played or processed, it is decompressed, and the exact data that was originally recorded is restored. In the case of a **lossy compression algorithm**, it's impossible to get back exactly the original data upon decompression. Examples of lossy compression formats are MP3, AAC, Ogg Vorbis, and the μ-law and A-law compression used in AU files. Examples of lossless compression algorithms include FLAC (Free Lossless Audio Codec), Apple Lossless, MPEG-4 ALS (Audio Lossless Coding), Monkey's Audio, and TTA (True Audio). More details of audio codecs are given in Section 5.2.1.

Aside:
Why are 52,920,000 bytes equal to about 50.5 MB? You might expect a megabyte to be 1,000,000 bytes, but in the realm of computers, things are generally done in powers of 2. Thus, we use the following definitions:

kilo = 2^{10} = 1024
mega = 2^{20} = 1,048,576

You should become familiar with the following abbreviations:

kilobits	kb	2^{10} bits
kilobytes	kB	2^{10} bytes
megabits	Mb	2^{20} bits
megabytes	MB	2^{20} bytes

Based on these definitions, 52,920,000 bytes is converted to megabytes by dividing by 1,048,576 bytes.

Unfortunately, usage is not entirely consistent. You'll sometimes see "kilo" assumed to be 1000 and "mega" assumed to be 1,000,000, e.g., in the specification of the storage capacity of a CD.

With the introduction of portable music players, copy-protected audio files became more prevalent. Apple introduced iTunes in 2001, allowing users to purchase and download music from their online store. The audio files, encoded in a proprietary version of the AAC format and using the *.m4p* file extension, were protected with Apple's FairPlay DRM system. DRM enforces limits on where the file can be played and whether it can be shared or copied. In 2009, Apple lifted restrictions on music sold from its iTunes store, offering an unprotected *.m4a* file as an alternative to *.m4p*. Copy-protection is generally embedded within container file formats like MP3. WMA (Windows Media Audio) files are another example, based on the Advanced Systems Format (ASF) and providing DRM support.

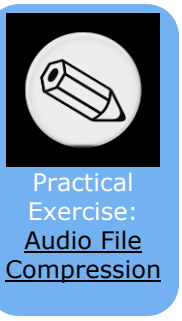
Common audio file types are summarized in Table 5.1.

File Type	Platform	File Extensions	Compression	Container	Proprietary	DRM
PCM	cross	.pcm	no	no	no	no
RAW	cross	.raw	no	no	no	no
WAV	cross	.wav	Optional (lossy)	yes, RIFF format	no	no
AIFF	Mac	.aif, .aiff,	no	yes, IFF format	no	no
AIFF-C	Mac	.aifc	yes, with various codecs (lossy)	yes, IFF format	no	
CAF	Mac	.caf	yes	yes	no	no
AU	Unix/Linux	.au, .snd	optional μ-law (lossy)	yes	no	no
MP3	cross	.mp3	MPEG (lossy)	yes	license required for distribution or sale of codec but not for use	optional
AAC	cross	.m4a, .m4b, .m4p, .m4v, .m4r, .3gp, .mp4, .aac	AAC (lossy)	more of a compression standard than a container; ADIF is container	license required for distribution or sale of codec but not for use	
WMA	Windows	.wma	WMA (lossy)	yes	yes	optional
OGG Vorbis	cross	.ogg, .oga	Vorbis (lossy)	yes	no, open source	optional
FLAC	cross	.flac	FLAC (lossless)	yes	no, open source	optional

Table 5.1 Common audio file types

AIFF and WAV have been the most commonly used file types in recent years. CAF files are an extension of AIFF files without AIFF's 4 GB size limit. This additional file size was needed for all the looping and other metadata used in GarageBand and Logic.

All along the way as you work with digital audio, you'll have to make choices about the format in which you save your files. A general strategy is this:



- When you're processing the audio in a software environment such as Audition, Logic, or Pro Tools, save your work in the software's proprietary format until you're finished working on it. These formats retain meta-information about non-destructive processes – filters, EQ, etc. – applied to the audio as it plays. Non-destructive processes do not change the original audio samples. Thus, they are easily undone, and you can always go back and edit the audio data in other ways for other purposes if needed.
- The originally recorded audio is the best information you have, so it's always good to keep an uncompressed copy of this. Generally, you should keep as much data as possible as you edit an audio file, retaining the highest bit depth and sampling rate appropriate for the work.
- At the end of processing, save the file in the format suitable for your platform of distribution (e.g., CD, DVD, web, or live performance). This may be compressed or uncompressed, depending on your purposes.

5.2 Applications

5.2.1 Choosing an Appropriate Sampling Rate

Before you start working on a project you should decide what sampling rate you're going to be working with. This can be a complicated decision. One thing to consider is the final delivery of your sound. If, for example, you plan to publish this content only as an audio CD, then you might choose to work with a 44,100 Hz sampling rate to begin with since that's the required sampling rate for an audio CD. If you plan to publish your content to multiple formats, you might choose to work at a higher sampling rate and then convert down to the rate required by each different output format.

The sampling rate you use is directly related to the range of frequencies you can sample. With a sampling rate of 44,100 Hz, the highest frequency you can sample is 22,050 Hz. But if 20,000 Hz is the upper limit of human hearing, why would you ever need to sample a frequency higher than that? And why do we have digital systems able to work at sampling rates as high as 192,000 Hz?

First of all, the 20,000 Hz upper limit of human hearing is an average statistic. Some people can hear frequencies higher than 20 kHz, and others stop hearing frequencies after 16 kHz. The people who can actually hear 22 kHz might appreciate having that frequency included in the recording. It is, however, a fact that there isn't a human being alive who can hear 96 kHz, so why would you need a 192 kHz sampling rate?

Perhaps we're not always interested in the range of human hearing. A scientist who is studying bats, for example, may not be able to hear the high frequency sounds the bats make but may need to capture those sounds digitally to analyze them. We also know that musical instruments generate harmonic frequencies much higher than 20 kHz. Even though you can't consciously hear those harmonics, their presence may have some impact on the harmonics you *can* hear. This might explain why someone with well-trained ears can hear the difference between a recording sampled at 44.1 kHz and the same recording sampled at 192 kHz.

The catch with recording at those higher sampling rates is that you need equipment capable of capturing frequencies that high. Most microphones don't pick up much above 22 kHz, so running the signal from one of those microphones into a 96 kHz ADC isn't going to give you any of the frequency benefits of that higher sampling rate. If you're willing to spend more money, you can get a microphone that can handle frequencies as high at 140 kHz. Then you need to make sure that every further device handling the audio signal is capable of working with and delivering frequencies that high.

If you don't need the benefits of sampling higher frequencies, the other reason you might choose a higher sampling rate is to reduce the latency of your digital audio system, as is discussed in Section 5.2.3.

A disadvantage to consider is that higher sampling rates mean more audio data, and therefore larger file sizes. Whatever your reasons for choosing a sampling rate, the important thing to remember is that you need to stick to that sampling rate for every audio file and every piece of equipment in your signal chain. Working with multiple sampling rates at the same time can cause a lot of problems.

5.2.2 Input Levels, Output Levels, and Dynamic Range

In this section, we consider how to set input and output levels properly and how these settings affect dynamic range.

When you get ready to make a digital audio recording or set sound levels for a live performance, you typically test the input level and set it so that your loudest inputs don't clip. **Clipping** occurs when the sound level is beyond the maximum input or output voltage level. It manifests itself as unwanted distortion or breaks in the sound. When capturing vocals, you can set the sound levels by asking a singer to sing the loudest note he thinks he's going to sing in the whole piece, and make sure that the level meter doesn't hit the limit. Figure 5.21 shows this situation in a software interface. The level meter at the bottom of the figure has hit the right hand side and thus has turned red, indicating that the input level is too high and clipping has occurred. In this case, you need to turn down the input level and test again before recording an actual take. The input level can be changed by a knob on your audio interface or, in the case of some advanced interfaces with digitally controlled analog preamplifiers, by a software interface accessible through your operating system. Figure 5.22 shows the input gain knob on an audio interface.

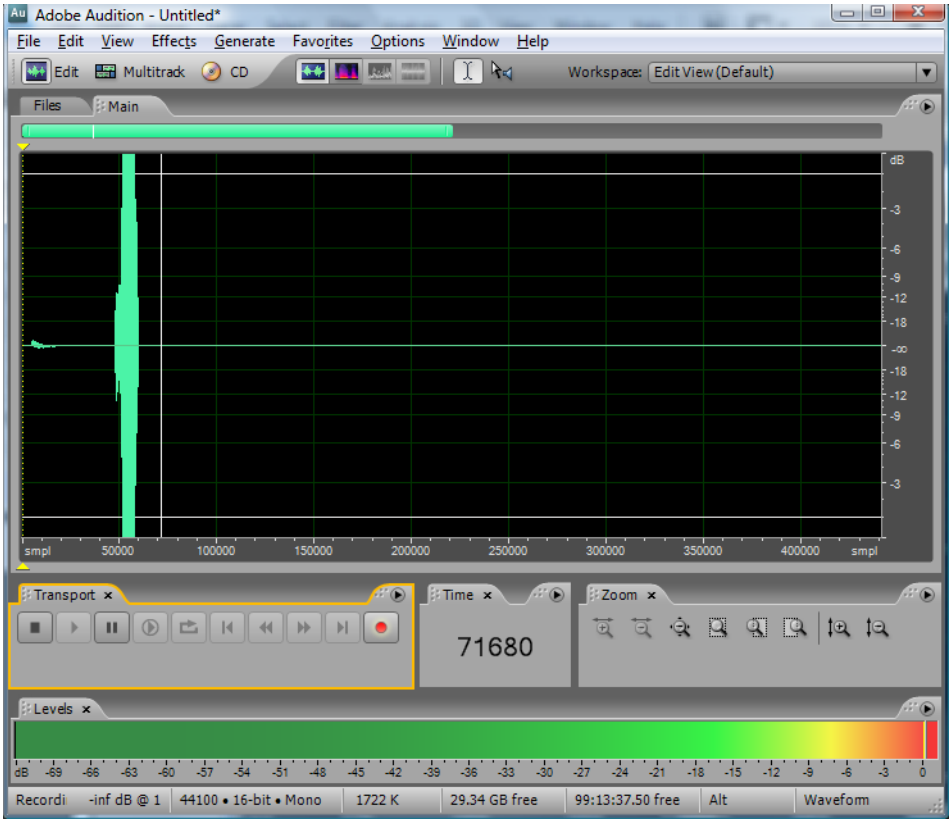


Figure 5.21 Clipped input level

Figure 5.22 Input gain knob on audio interface

Let's look more closely at what's going on when you set the input level. Any hardware system has a maximum input voltage. When you set the input level for a recording session or live performance, you're actually adjusting the analog input amplifier for the purpose of ensuring that the loudest sound you intend to capture does not generate a voltage higher than the maximum allowed by the system. However, when you set input levels, there's no guarantee that the singer won't sing louder than expected. Also, depending on the kind of sound you're capturing, you might have **transients** – short loud bursts of sound like cymbal claps or drum beats – to account for. When setting the input level, you need to save some headroom for these occasional loud sounds. **Headroom** is loosely defined as the distance between your "usual" maximum amplitude and the amplitude of the loudest sound that can be captured without clipping. Allowing for sufficient headroom obviously involves some guesswork. There's no guarantee that the singer won't sing louder than expected, or some unexpectedly loud transients may occur as you record, but you make your best estimate for the input level and adjust later if necessary, though you might lose a good take to clipping if you're not careful.

Let's consider the impact that the initial input level setting has on the dynamic range of a recording. Recall from Section 5.1.2.4 that the quietest sound you can capture is relative to the loudest as a function of the bit depth. A 16-bit system provides a dynamic range of approximately 96 dB. This implies that, in the absence of environment noise, the quietest sounds that you can capture are about 96 dB below the loudest sounds you can capture. That 96 dB value is also assuming that you're able to capture the loudest sound at the exact maximum input amplitude without clipping, but as we know leaving some headroom is a good idea. The quietest sounds that you can capture lie at what is called the **noise floor**. We could look at the noise floor from two directions, defining it as either the minimum amplitude level that can be captured or the maximum amplitude level of the noise in the system. With no environment or system noise, the noise floor is determined entirely by the bit depth, the only noise being quantization error.

In the software interface shown in Figure 5.21, input levels are displayed in decibels-full-scale (dBFS). The audio file shown is a sine wave that starts at maximum amplitude, 0 dBFS, and fades all the way out. (The sine wave is at a high enough frequency that the display of the full timeline renders it as a solid shape, but if you zoom in you see the sine wave shape.) Recall that with dBFS, the maximum amplitude of sound for the given system is 0 dBFS, and increasingly negative numbers refer to increasingly quiet sounds. If you zoom in on this waveform and look at the last few samples (as shown in the bottom portion of Figure 5.23), you can see that the lowest sample values – the ones at the end of the fade – are –90.3 dBFS. This is the noise floor resulting from quantization error for a bit depth of 16 bits. A noise floor of –90.3 dBFS implies that any sound sample that is more than 90.3 dB below the maximum recordable amplitude is recorded as silence.

Aside:
Why is the smallest value for a 16-bit sample –90.3 dB? Because
$20\ log_{10}\left(\frac{1}{32768}\right) = 90.3\ dB$.

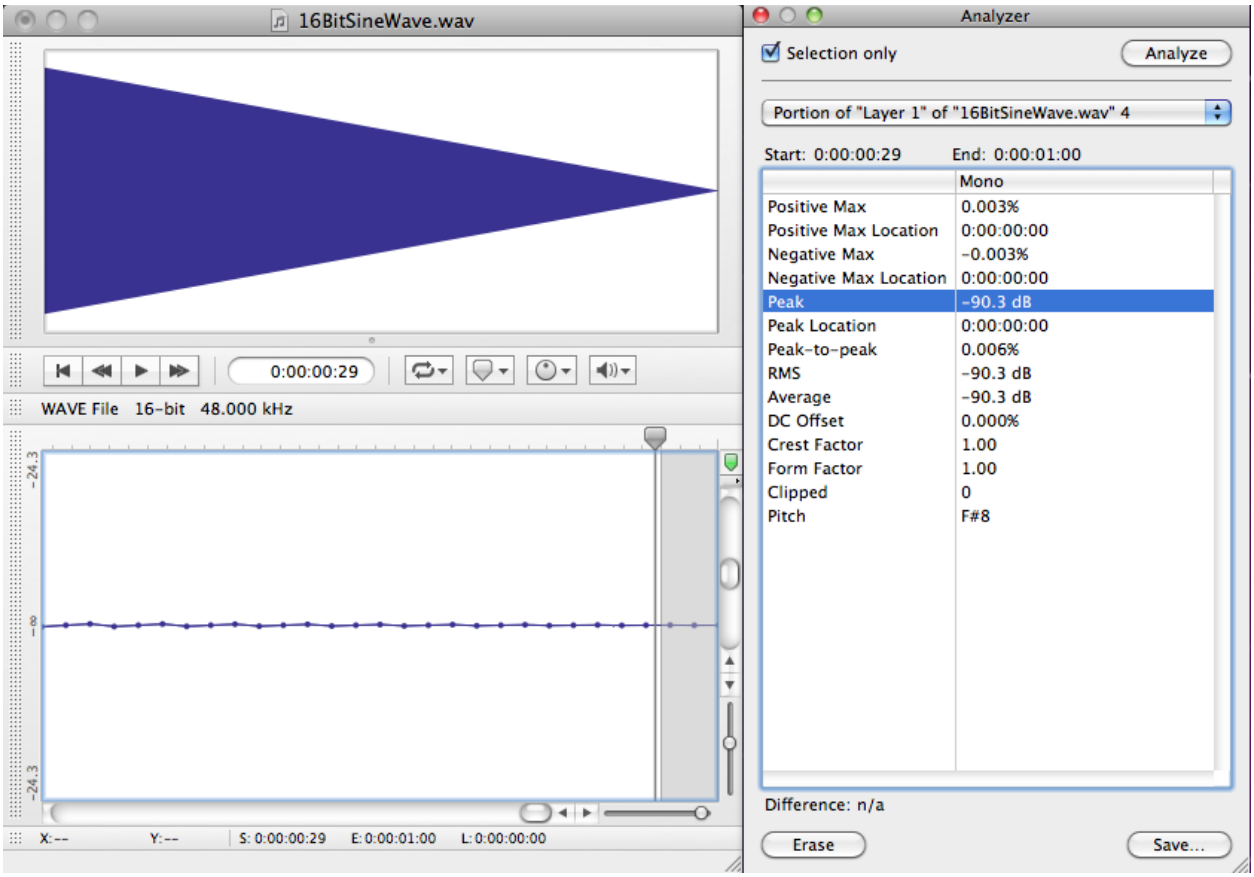


Figure 5.23 Finding the smallest possible samples in a digital audio recording

In reality, there is almost always some real-world noise in a sound capturing system. Every piece of audio hardware makes noise. For example, noise can arise from electrical interference on long audio cables or from less-than-perfect audio detection in microphones. Also, the environment in which you're capturing the sound will have some level of background noise such as from the air conditioning system. The maximum amplitude of the noise in the environment or the sound-capturing system constitutes the real noise floor. Sounds below this level are masked by the noise. This means that either they're muddied up with the noise, or you can't distinguish them at all as part of the desired audio signal. In the presence of significant environmental or system noise during recording, the available dynamic range of a 16-bit recording is the difference between 0 dBFS and the noise floor caused by the environment and system noise. For example, if the noise floor is –70 dBFS, then the dynamic range is 70 dB. (Remember that when you subtract dBFS from dBFS, you get dB.)

So we've seen that the bit depth of a recorded audio file puts a fixed limit on the available dynamic range, and that this potential dynamic range can be made smaller by environmental noise. Another thing to be aware of is that you can waste some of the available dynamic range by setting your input levels in a way that leaves more headroom than you need. If you have 96 dB of dynamic range available but it turns out that you use only half of it, you're squeezing your actual sound levels into a smaller dynamic range than necessary. This results in less accurate quantization than you could have had, and it puts more sounds below the noise floor than would have been there if you had used a greater part of your available dynamic range. Also, if you

underuse your available dynamic range, you might run into problems when you try to run any long fades or other processes affecting amplitude, as demonstrated in the practical exercise "Bit Depth and Dynamic Range" linked in this section.

It should be clarified that increasing the input levels also increases any background environmental noise levels captured by a microphone, but can still benefit the signal by boosting it higher above any electronic or quantization noise that occurs downstream in the system. The only way to get better dynamic range over your air conditioner hum is to turn it off or get the microphone closer to the sound you want to record. This increases the level of the sound you want without increasing the level of the background noise.


Practical Exercise: Bit Depth and Dynamic Range

So let's say you're recording with a bit depth of 16 and you've set your input level just about perfectly to use all of that dynamic range possible in your recording. Will you actually be able to get the benefit of this dynamic range when the sound is listened to, considering the range of human hearing, the range of the sound you want to hear, and the background noise level in a likely listening environment? Let's consider the *dynamic range of human hearing* first and the *dynamic range of the types of things we might want to listen to*. Though the human ear can technically handle a sound as loud as 120 dBSPL, such high amplitudes certainly aren't comfortable to listen to, and if you're exposed to sound at that level for more than a few minutes, you'll damage your hearing. The sound in a live concert or dance club rarely exceeds 100 dBSPL, which is pretty loud to most ears. Generally, you can listen to a sound comfortably up to about 85 dBSPL. The quietest thing the human ear can ear is just above 0 dBSPL. The dynamic range between 85 dBSPL and 0 dBSPL is 85 dB. Thus, the 96 dB dynamic range provided by 16-bit recording effectively pushes the noise floor below the threshold of human hearing at a typical listening level.

We haven't yet considered the *noise floor of the listening environment*, which is defined as the maximum amplitude of the unwanted background noise in the listening environment. The average home listening environment has a noise floor of around 50 dBSPL. With the dishwasher running and the air conditioner going, that noise floor could get up to 60 dBSPL. In a car (perhaps the most hostile listening environment) you could have a noise floor of 70 dBSPL or higher. Because of this high noise floor, car radio music doesn't require more than about 25 dB of dynamic range. Does this imply that the recording bit depth should be dropped down to eight bits or even less for music intended to be listened to on the car radio? No, not at all.

Here's the bottom line. You'll almost always want to do your recordings in 16 bit sample sizes, and sometimes 24 bits or 32 bits are even better, even though there's no listening environment on earth (other than maybe an anechoic chamber) that allows you the benefit of the dynamic range these bit depths provide. The reason for the large bit depths has to do with the processing you do on the audio before you put it into its final form. Unfortunately, you don't always know how loud things will be when you capture them. If you guess low when setting your input level, you could easily get into a situation where most of the audio signal you care about is at the extreme quiet end of your available dynamic range, and fadeouts don't work well because the signal too quickly fades below the noise floor. In most cases, a simple sound check and a bit of pre-amp tweaking can get you lined up to a place where 16 bits are more than enough. But if you don't have the time, if you'll be doing lots of layering and audio processing, or if you just can't be bothered to figure things out ahead of time, you'll probably want to use 24 bits. Just keep in mind that the higher the bit depth, the larger the audio files are on your computer.

An issue we're not considering in this section is applying dynamic range compression as one of the final steps in audio processing. We mentioned that the dynamic range of car radio music's listening environment is about 25 dB. If you play music that covers a wider dynamic range than 25 dB on a car radio, a lot of the soft parts are going to be drowned out by the noise caused by tire vibrations, air currents, etc. Turning up the volume on the radio isn't a good solution, because it's likely that you'll have to make the loud parts uncomfortably loud in order to hear the quiet parts. In fact, the dynamic range of music prepared for radio play is often compressed after it has been recorded, as one of the last steps in processing. It might also be further compressed by the radio broadcaster. The dynamic range of sound produced for theatre can be handled in the same way, its dynamic range compressed as appropriate for the dynamic range of the theatre listening environment. Dynamic range compression is covered in Chapter 7.

5.2.3 Latency and Buffers

In Section 5.1.4, we looked at the digital audio signal path during recording. A close look at this signal path shows how delays can occur in between input and output of the audio signal, and how such delays can be minimized.


Flash Tutorial: Reducing Latency

Latency is the period of time between when an audio signal enters a system and when the sound is output and can be heard. Digital audio systems introduce latency problems not present in analog systems. It takes time for a piece of digital equipment to process audio data, time that isn't required in fully analog systems where sound travels along wires as electric current at nearly the speed of light. An immediate source of latency in a digital audio system arises from analog-to-digital and digital-to-analog conversions. Each conversion adds latency on the order of milliseconds to your system. Another factor influencing latency is buffer size. The input buffer must fill up before the digitized audio data is sent along the audio stream to output. Buffer sizes vary by your driver and system, but a size of 1024 samples would not be usual, so let's use that as an estimate. At a sampling rate of 44.1 kHz , it would take about 23 ms to fill a buffer with 1024 samples, as shown below.

$$\frac{1\ sec}{44{,}100\ samples} * 1024\ samples \approx 0.023\ s = 23\ ms$$

Thus, total latency including the time for ADC, DAC, and buffer-filling is on the order of milliseconds. A few milliseconds of delay may not seem very much, but when multiple sounds are expected to be synchronized when they arrive at the listener, this amount of latency can be a problem, resulting in phase offsets and echoes.

Let's consider a couple of scenarios in which latency can be a problem, and then look at how the problem can be dealt with. Imagine a situation where a singer is singing live on stage. Her voice is taken in by the microphone and undergoes digital processing before it comes out the loudspeakers. In this case, the sound is not being recorded, but there's latency nonetheless. Any ADC/DAC conversions and audio processing along the signal path can result in an audible delay between when a singer sings into a microphone and when the sound from the microphone radiates out of a loudspeaker. In this situation, the processed sound arrives at the audience's ears after the live sound of the singer's voice, resulting in an audible echo. The simplest way to reduce the latency here is to avoid analog-to-digital and digital-to-analog conversions whenever possible. If you can connect two pieces of digital sound equipment using a digital signal

transmission instead of an analog transmission, you can cut your latency down by at least two milliseconds because you'll have eliminated two signal conversions.

Buffer size contributes to latency as well. Consider a scenario in which a singer's voice is being digitally recorded (Figure 5.20). When an audio stream is captured in a digital device like a computer, it passes through an input buffer. This input buffer must be large enough to hold the audio samples that are coming in while the CPU is off somewhere else doing other work. When a singer is singing into a microphone, audio samples are being collected at a fixed rate – say 44,100 samples per second. The singer isn't going to pause her singing and the sound card isn't going to slow down the number of samples it takes per second just because the CPU is busy. If the input buffer is too small, samples have to be dropped or overwritten because the CPU isn't there in time to process them. If the input buffer is sufficiently large, it can hold the samples that accumulate while the CPU is busy, but the amount of time it takes to fill up the buffer is added to the latency.

The singer herself will be affected by this latency is she's listening to her voice through headphones as her voice is being digitally recorded (called **live sound monitoring**). If the system is set up to use **software monitoring,** the sound of the singer's voice enters the microphone, undergoes ADC and then some processing, is converted back to analog, and reaches the singer's ears through the headphones. Software monitoring requires one analog-to-digital and one digital-to-analog conversion. Depending on the buffer size and amount of processing done, the singer may not hear herself back in the headphones until 50 to 100 milliseconds after she sings. Even an untrained ear will perceive this latency as an audible echo, making it extremely difficult to sing on beat. If the singer is also listening to a backing track played directly from the computer, the computer will deliver that backing track to the headphones sooner than it can deliver the audio coming in live to the computer. (A backing track is a track that has already been recorded and is being played while the singer sings.)


Flash Tutorial: Direct Monitoring

Latency in live sound monitoring can be avoided by **hardware monitoring** (also called **direct monitoring)**. Hardware monitoring splits the newly digitized signal before sending it into the computer, mixing it directly into the output and eliminating the longer latencies caused by analog-to-digital conversion and input buffers (Figure 5.25). The disadvantage of hardware monitoring is that the singer cannot hear her voice with processing such as reverb applied. (Audio interfaces that offer direct hardware monitoring with zero-latency generally let you control the mix of what's coming directly from the microphone and what's coming from the computer. That's the purpose of the **monitor mix knob**, circled in red in Figure 5.24.) When the mix knob is turned fully counterclockwise, only the direct input signals (e.g., from the microphone) are heard. When the mix knob is turned fully counterclockwise, only the signal from the DAW software is heard.

Figure 5.24 Mix knob on audio interface

In general, the way to reduce latency caused by buffer size is to use the most efficient driver available for your system. In Windows systems, ASIO drivers are a good choice. ASIO drivers cut down on latency by allowing your audio application program to speak directly to the sound card, without having to go through the operating system. Once you have the best driver in place, you can check the interface to see if the driver gives you any control over the buffer size. If you're allowed to adjust the size, you can find the optimum size mostly by trial and error. If the buffer is too large, the latency will be bothersome. If it's too small, you'll hear breaks in the audio because the CPU may not be able to return quickly enough to empty the buffer, and thus audio samples are dropped out.

With dedicated hardware systems (digital audio equipment as opposed to a DAW based on your desktop or laptop computer) you don't usually have the ability to change the buffer size because those buffers have been fixed at the factory to match perfectly the performance of the specific components inside that device. In this situation, you can reduce the latency of the hardware by increasing your internal sampling rate. This may seem counterintuitive at first because a higher sampling rate means that you're processing more data per second. This is true, but remember that the buffer sizes have been specifically set to match the performance capabilities of that hardware, so if the hardware gives you an option to run at a higher sampling rate, you can be confident that the system is capable of handling that speed without errors or dropouts. For a buffer of 1024 samples, a sampling rate of 192 kHz has a latency of about 5.3 ms, as shown below.

$$1024\ samples * \frac{1\ sec}{192{,}000\ samples} \approx 5.3\ ms$$

If you can increase your sampling rate, you won't necessarily get a better sound from your system, but the sound is delivered with less latency.

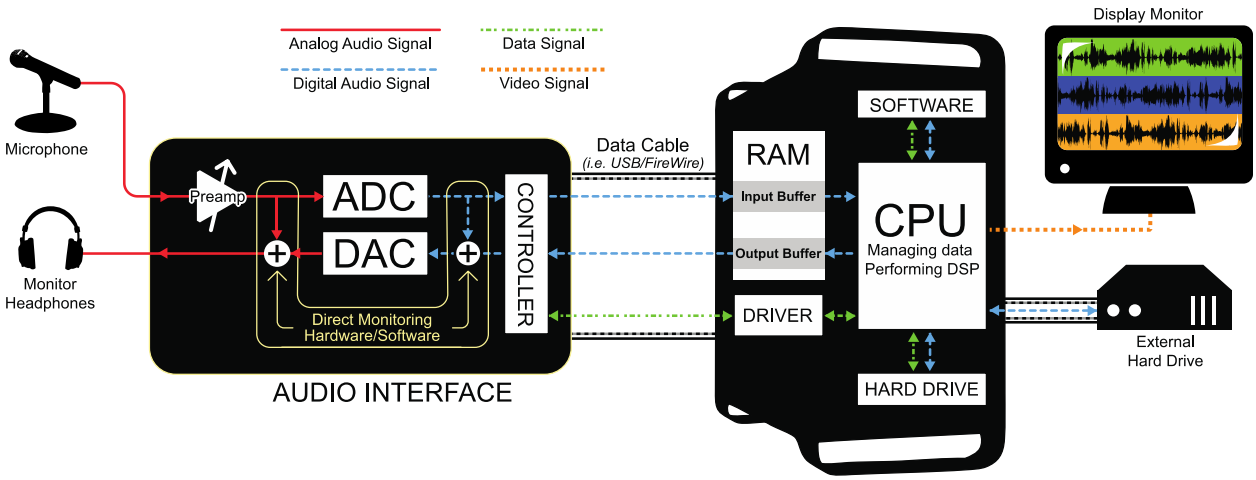


Figure 5.25 Signal path in digital audio recording with direct monitoring

5.2.4 Word Clock

When transmitting audio signals between digital audio hardware devices, you need to decide whether to transmit in digital or analog format. Transmitting in analog involves performing analog-to-digital conversions coming into the devices and digital-to-analog conversions coming out of the devices. As described in Section 5.2.1, you pay for this strategy with increased latency in your audio system. You may also pay for this in a loss of quality in your audio signal as a result of multiple quantization errors and a loss of frequency range if each digital device is using a different sampling rate. If you practice good gain structure (essentially, controlling amplitude changes from one device to the next) and keep all your sampling rates consistent, the loss of quality is minimal, but it is still something to consider when using analog interconnects. (See Chapter 8 for more on setting gain structure.)

Interconnecting these devices digitally can remove the latency and potential signal loss of analog interconnects, but digital transmission introduces a new set of problems, such as timing. There are several different digital audio transmission protocols, but all involve essentially the same basic process in handling data streams. The signal is transmitted as a stream of small blocks or frames of data containing the audio sample along with timing, channel information, and error correction bits. These data blocks are a constant stream of bits moving down a cable. The stream of bits is only meaningful when it gets split back up into the blocks containing the sample data in the same way it was sent out. If the stream is split up in the wrong place, the data block is invalid. To solve this problem, each digital device has a clock that runs at the speed of the sampling rate defined for the audio stream. This clock is called a **word clock.** Every time the clock ticks, the digital device grabs a new block of data – sometimes called an **audio word** – from the audio stream. If the device receiving the digital audio stream has a word clock that is running in sync with the word clock of the device sending the digital audio stream, each block that is transmitted is received and interpreted correctly. If the word clock of the receiving devices falls out of sync, it starts chopping up the blocks in the wrong place and the audio data will be invalid.

Even the most expensive word clock circuit is imperfect. This imperfection is measured in parts per million (ppm), and can be up to 50 ppm even in good quality equipment. At a

sampling rate of 44.1 kHz, this equates to a potential drift of $44100 * \frac{50}{1000000} \approx 2.2$ samples per second. Even if two word clocks start at precisely the same time, they are likely to drift at different rates and thus will eventually be out of sync. To avoid the errors that result from word clock drift, you need to synchronize the word clocks of all your digital devices. There are three basic strategies for word clock synchronization. The strategy you choose depends on the capability of the equipment you are using.

The first word clock synchronization strategy is to slave all your digital devices to a dedicated word clock generator. Any time you can go with a dedicated hardware solution, chances are good that the hardware is going to be pretty reliable. If the box only has to do one thing, it will probably be able to do it well. A dedicated word clock generator has a very stable word clock signal and several output connectors to send that word clock signal to all the devices in your system. It may also have the ability to generate and sync to other synchronization signals such as MIDI Time Code (MTC), Linear Time Code (LTC), and video black burst (the word clock equivalent for video equipment). An example of a dedicated synchronization tool is shown in Figure 5.26.

Figure 5.26 - This is a dedicated synchronization tool from Avid

External word clock synchronization is typically accomplished using low impedance coaxial cable with BNC connectors. If your word clock generator has several outputs, you can connect each device directly to the clock master as shown in Figure 5.27. Otherwise you can connect the devices up in sequence from a single word clock output of your clock master shown in Figure 5.28.

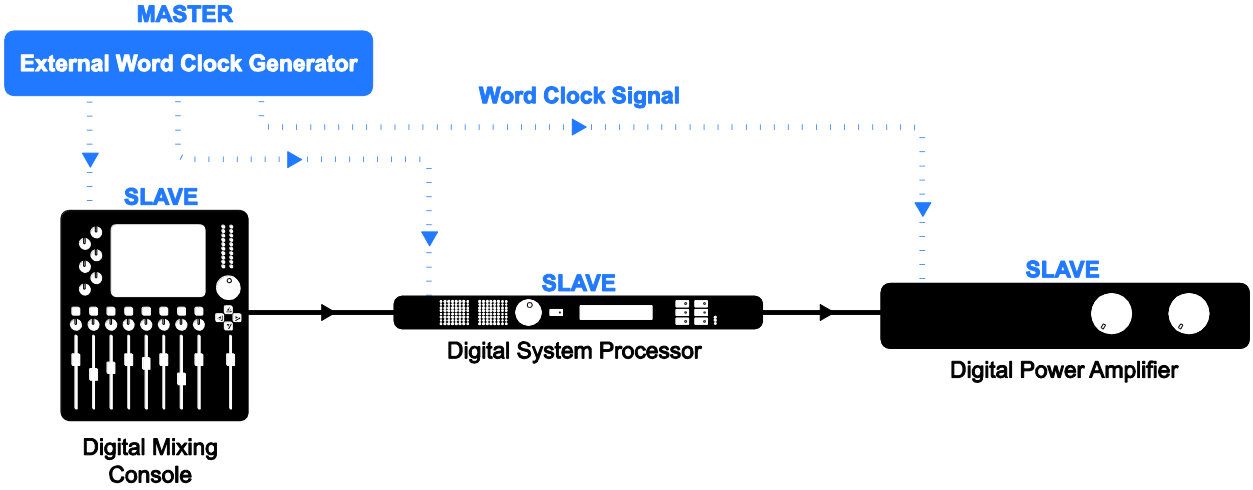


Figure 5.27 - Synchronizing to an external word clock generator with multiple outputs

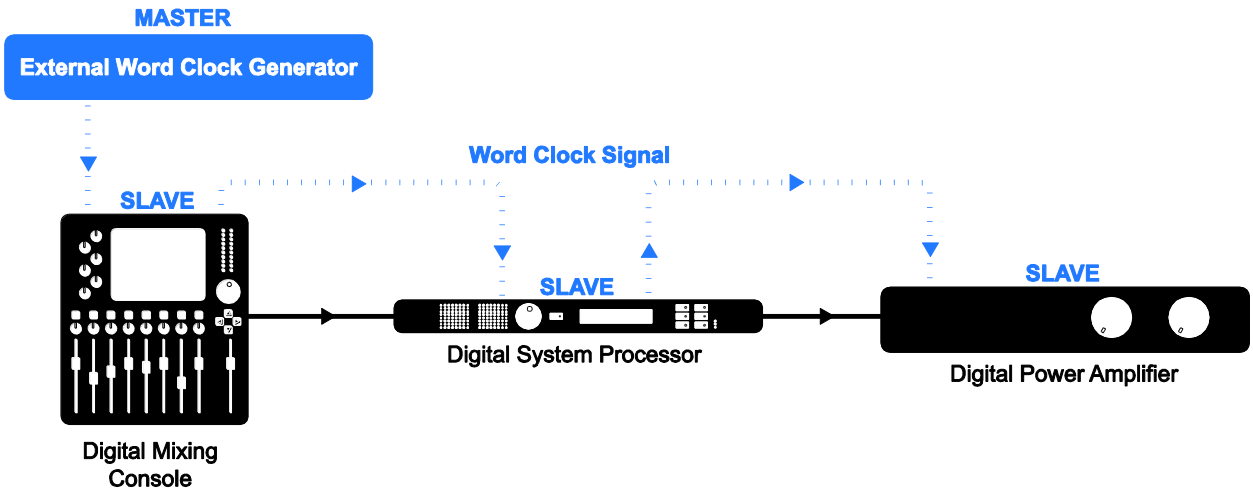


Figure 5.28 - Synchronizing to an external word clock generator with a single output

If you don't have a dedicated word clock generator, you could choose one of the digital audio devices in your system to be the word clock master and slave all the other devices in your system to that clock using the external word clock connections as shown in Figure 5.29. The word clock of the device in your system is probably not as stable as a dedicated word clock generator (in that it may have a small amount of jitter), but as long as all the devices are following that clock, you will avoid any errors.

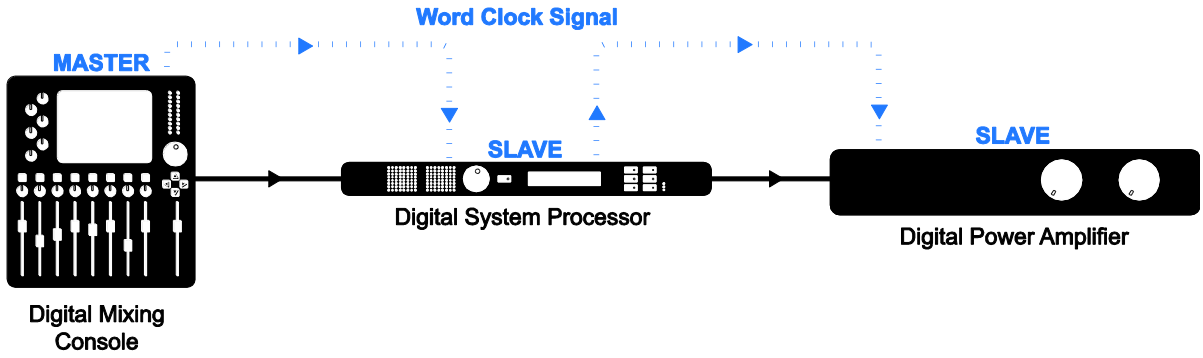


Figure 5.29 - Synchronizing to the first digital device in your signal chain using external word clock connections

In some cases your equipment may not have external word clock inputs. This is common in less expensive equipment. In that situation you could go with a self-clocking solution where the first device in your signal chain is designated as the word clock master and each device in the signal chain is set to slave to the word clock signal embedded in the audio stream coming from the previous device in the signal chain, as shown in Figure 5.30.

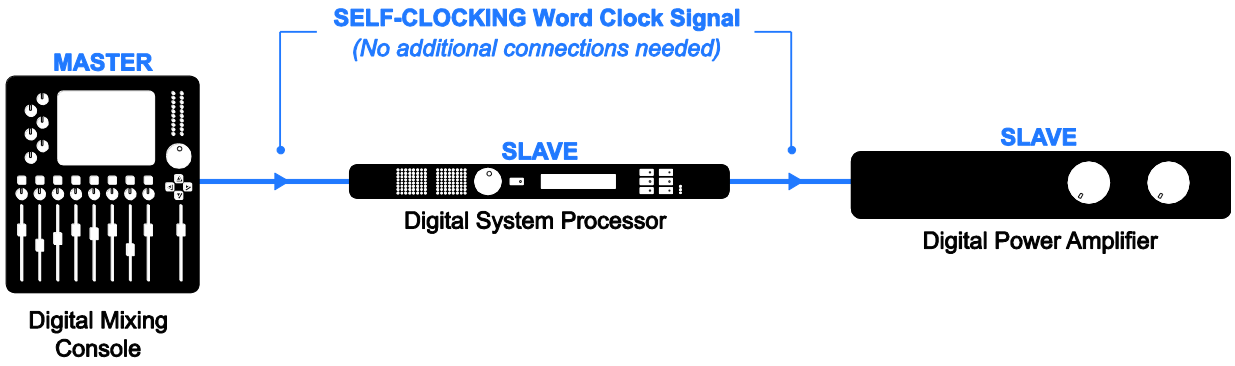


Figure 5.30 - Synchronizing to the clocking signal embedded in the digital audio stream

Regardless of the synchronization strategy you use, the goal is to have one word clock master with every other digital device in your system set to slave to the master word clock. If you set this up correctly, you should have no problems maintaining a completely digital signal path through your audio system and benefit from the decreased latency from input to output.

5.3 Science, Mathematics, and Algorithms

5.3.1 Reading and Writing Audio Files in MATLAB

A number of exercises in this book ask that you experiment with audio processing by reading an audio file into MATLAB or a C++ program, manipulate it in some way, and either listen directly to the result or write the result back to a file. In Chapter 2 we introduced how to read in WAV files in MATLAB. Let's look at this in more detail now.

In some cases, you may want to work with *raw* audio data coming from an external source. One way to generate raw audio data is to take a clip of music or sound and, in Audacity or some other higher level tool, save or export the clip as an uncompressed or raw file. Audacity allows you to save in uncompressed WAV or AIFF formats (Figure 5.31).

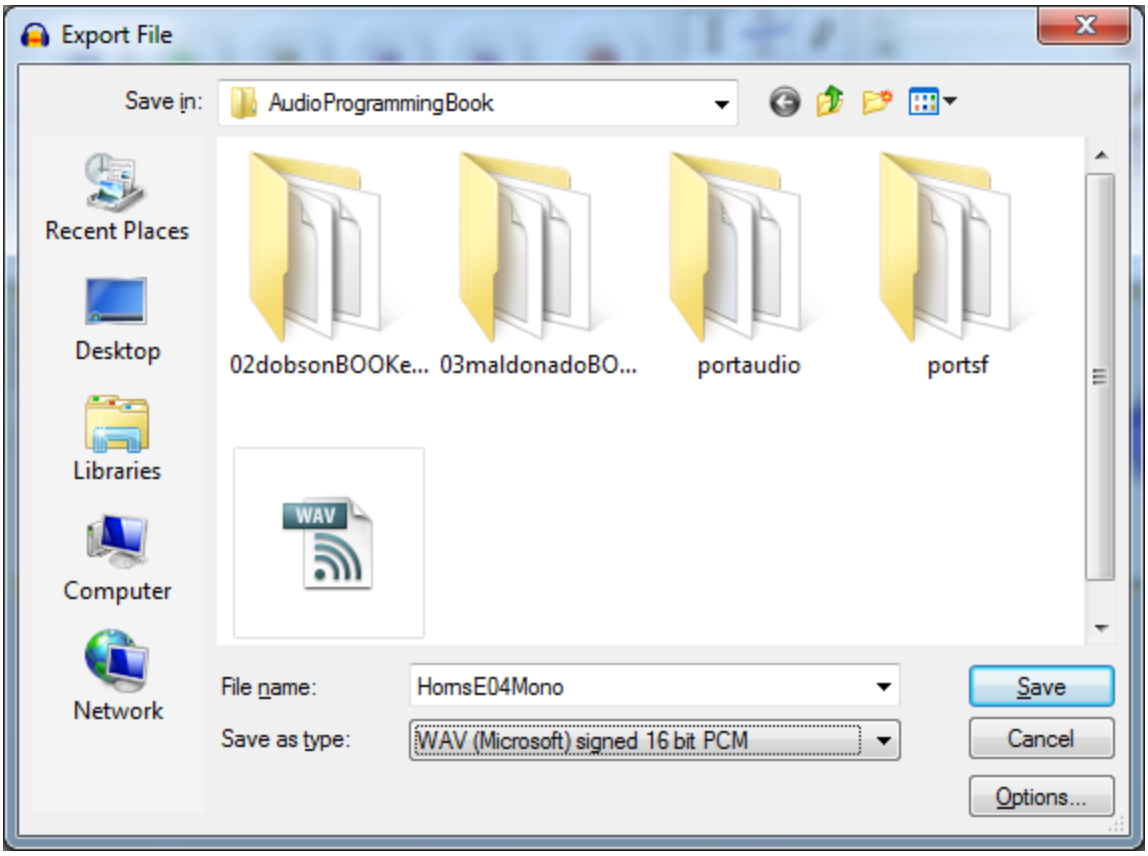


Figure 5.31 Exporting an uncompressed audio file in Audacity

From the next input box that pops open, you can see that although this file is being saved as uncompressed PCM, it still has a WAV header on it containing metadata.

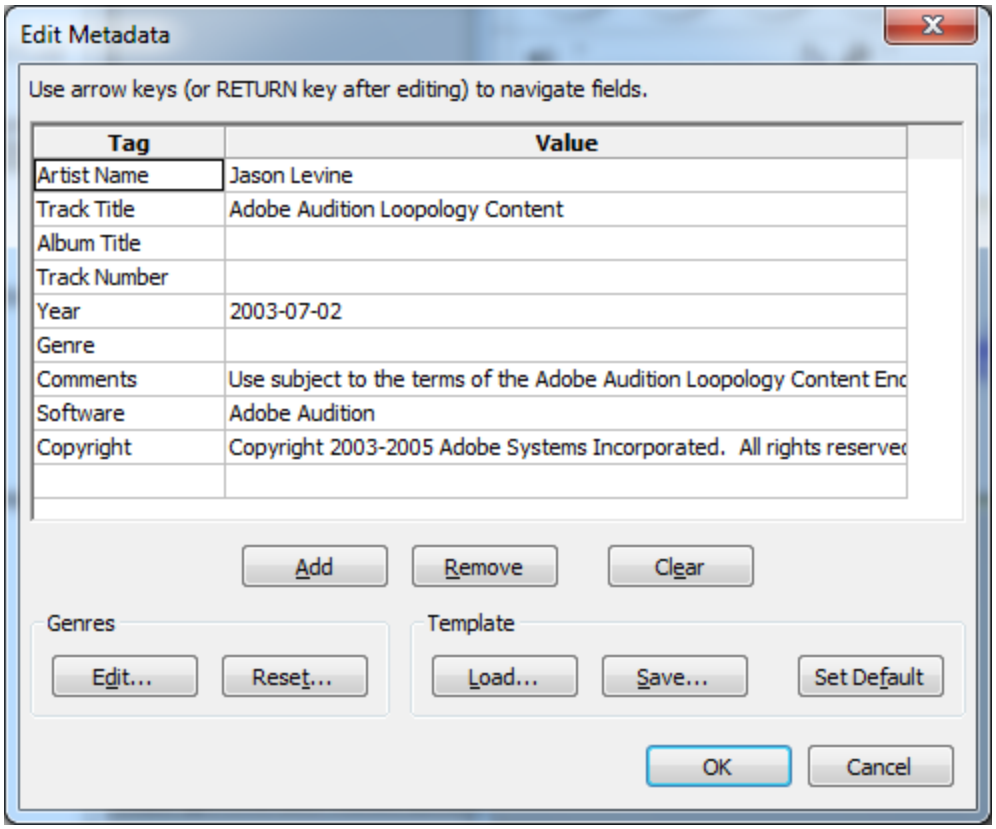


Figure 5.32 Prompt for metadata to be stored in header of uncompressed WAV file

The WAV file can be read an array in MATLAB with the following:

```
xWav = wavread('HornsE04Mono.wav');
```

The *wavread* function strips the header off and places the raw audio values into the array *x*. These values have a maximum range from –1 to 1. If you want to know the sampling rate *sr* and bit depth *b*, you can use this:

```
[xWav, sr, b] = wavread('HornsE04Mono.wav');
```

If the file is stereo, you'll get a two-dimensional array with the same number of samples in each channel.

Adobe Audition allows you to save an audio file as raw data with no header by deselecting the "Save extra non-audio information" box (Figure 5.33).

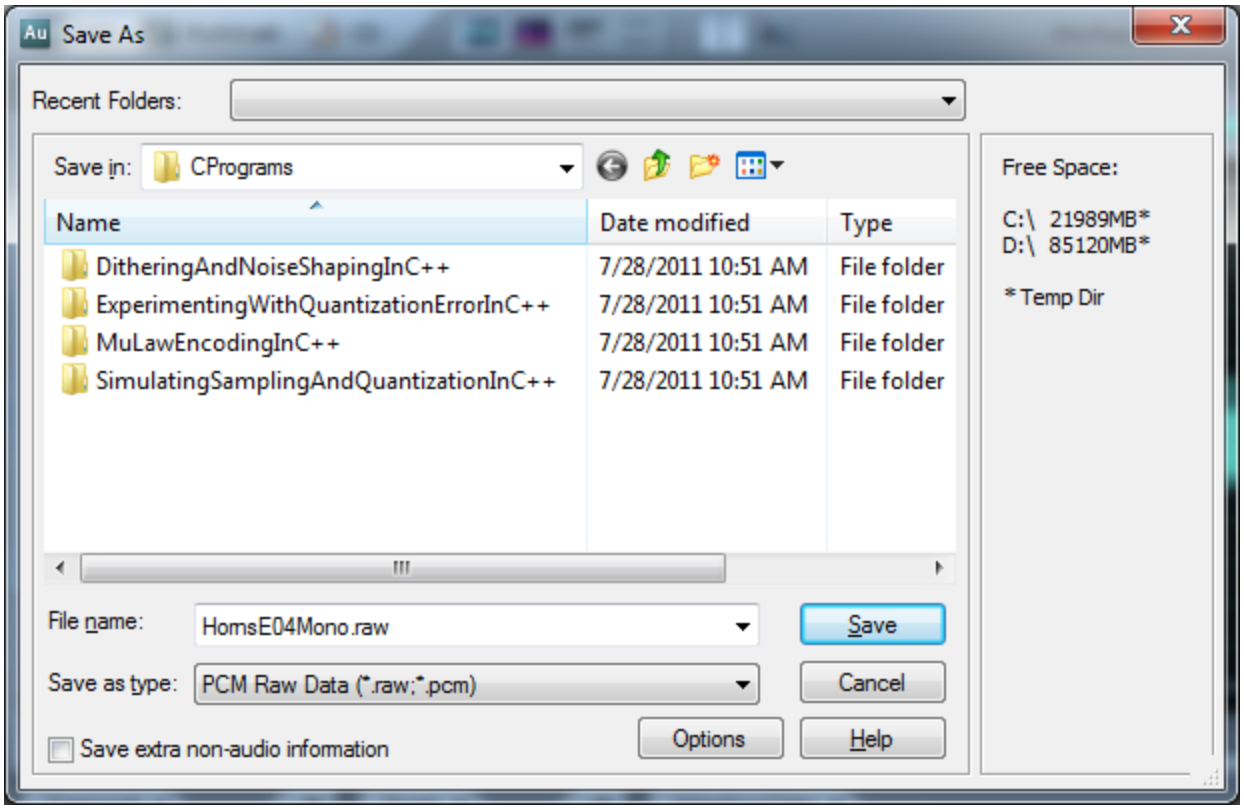


Figure 5.33 Saving a RAW audio file in Adobe Audition

When you save a file in this manner, you need to remember the sampling rate and bit depth. A raw audio file is read in MATLAB as follows:

```
fid = fopen('HornsE04Mono.raw', 'r');
xRaw16 = fread(fid, 'int16');
```

fid is the file handle, and the *r* in single quotes means that the file is being opened to be read. The type specifier *int16* is used in *fread* so that MATLAB knows to interpret the input as 16-bit signed integers. MATLAB's workspace window should show you that the values are in the maximum range of –32768 to 32767 (-2^{b-1} *to* $2^{b-1}-1$) (Figure 5.34). The values don't span the maximum range possible because they were at low amplitude to begin with.

For *HornsE04Mono_8bits.raw,* an 8-bit file, you can use

```
fid2 = fopen('HornsE04Mono_8bits.raw', 'r');
xRaw8 = fread(fid2, 'int8');
```

The values of *xRaw8* range from –128 to 127, as shown in Figure 5.34.

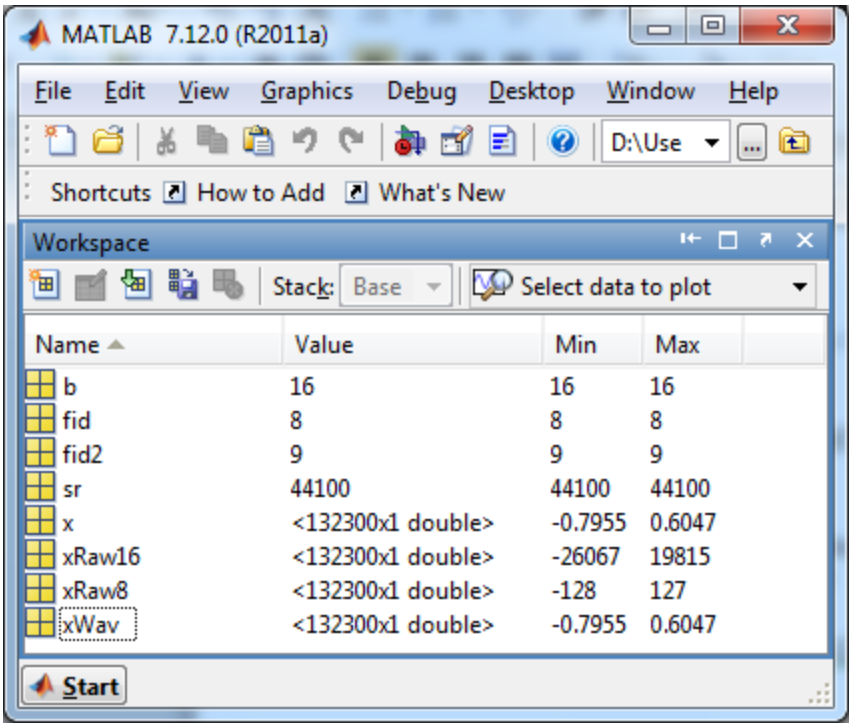


Figure 5.34 Values of variables after reading in audio files in MATLAB

A stereo RAW file has twice as many samples as the corresponding mono file, with the samples for the two stereo channels interleaved.

You can use the *fwrite* function in MATLAB to write the file back out to permanent storage after you've manipulated the values.

5.3.2 Raw Audio Data in C++

In Chapters 2 and 3, we showed you how to generate sinusoidal waveforms as arrays of values that could be sent directly to the sound device and played. But what if you want to work with more complicated sounds and music in C++? In this case, you'll need to be able to read audio files into your programs in order to experiment with audio processing.

In C++, raw audio data in 8-bit or 16-bit format can be read into an array of characters (*char*) or short integers (*short*), respectively, assuming that characters are stored in 8 bits and short integers are 16 bits on your system. This is demonstrated in Program 5.1. You can use this in the programming exercise on dithering and mu-law encoding.

```
//This program runs under OSS
//The program expects an 8-bit raw sound file.
//You can alter it to read a 16-bit file into short ints
#include <sys/ioctl.h> //for ioctl()
#include <math.h> //sin(), floor()
#include <stdio.h> //perror
#include <fcntl.h> //open
#include <linux/soundcard.h> //SOUND_PCM*
#include <stdlib.h>
#include <unistd.h>
```

```
using namespace std;

#define TYPE char
#define RATE 44100 //sampling rate
#define SIZE sizeof(TYPE) //size of sample, in bytes
#define CHANNELS 1 //number of stereo channels
#define PI 3.14159
#define SAMPLE_MAX (pow(2,SIZE*8 - 1) - 1)

void writeToSoundDevice(TYPE* buf, int deviceID, int buffSize) {
  int status;
  status = write(deviceID, buf, buffSize);
  if (status != buffSize)
    perror("Wrote wrong number of bytes\n");
  status = ioctl(deviceID, SOUND_PCM_SYNC, 0);
  if (status == -1)
    perror("SOUND_PCM_SYNC failed\n");
}

int main(int argc, char* argv[])
{
  int deviceID, arg, status, i, numSamples;
  numSamples = atoi(argv[1]);

  TYPE* samples = (TYPE *) malloc((size_t) numSamples * sizeof(TYPE)*
CHANNELS);
  FILE *inFile = fopen(argv[2], "rb");
  fread(samples, (size_t)sizeof(TYPE), numSamples*CHANNELS, inFile);
  fclose(inFile);

  deviceID = open("/dev/dsp", O_WRONLY, 0);
  if (deviceID < 0)
    perror("Opening /dev/dsp failed\n");
  arg = SIZE * 8;
  status = ioctl(deviceID, SOUND_PCM_WRITE_BITS, &arg);
  if (status == -1)
    perror("Unable to set sample size\n");
  arg = CHANNELS;
  status = ioctl(deviceID, SOUND_PCM_WRITE_CHANNELS, &arg);
  if (status == -1)
    perror("Unable to set number of channels\n");
  arg = RATE;
  status = ioctl(deviceID, SOUND_PCM_WRITE_RATE, &arg);
  if (status == -1)
    perror("Unable to set number of bits\n");

  writeToSoundDevice(samples, deviceID, numSamples * CHANNELS);
  FILE *outFile = fopen(argv[3], "wb");
  fwrite(samples, 1, numSamples*CHANNELS, outFile);
  fclose(outFile);
  close(deviceID);
}
```

Program 5.1 Reading raw data in a C++ program, OSS library

When you write a low-level sound program such as this, you need to use a sound library such as OSS or ALSA. There's a lot of documentation online that will guide you through installation of the sound libraries into the Linux environment and show you how to program with the libraries. (See the.websites cited in the references at the end of the chapter.) The ALSA version of Program 5.1 is given in Program 5.2.

```
/*This program demonstrates how to read in a raw
  data file and write it to the sound device to be played.
  The program uses the ALSA library.
  Use option -lasound on compile line.*/

#include </usr/include/alsa/asoundlib.h>
#include <math.h>
#include <iostream>
using namespace std;

static char *device = "default";    /*default playback device */
int main(int argc, char* argv[])
{
  int err, numSamples;
  snd_pcm_t *handle;
  snd_pcm_sframes_t frames;
  numSamples = atoi(argv[1]);
  char* samples = (char*) malloc((size_t) numSamples);
  FILE *inFile = fopen(argv[2], "rb");
  int numRead = fread(samples, 1, numSamples, inFile);
  fclose(inFile);
  if ((err=snd_pcm_open(&handle, device, SND_PCM_STREAM_PLAYBACK, 0)) < 0){
    printf("Playback open error: %s\n", snd_strerror(err));
    exit(EXIT_FAILURE);
  }
  if ((err =
   snd_pcm_set_params(handle,SND_PCM_FORMAT_U8,
     SND_PCM_ACCESS_RW_INTERLEAVED,1,44100, 1, 100000) ) < 0 ){
        printf("Playback open error: %s\n", snd_strerror(err));
        exit(EXIT_FAILURE);
   }
  frames = snd_pcm_writei(handle, samples, numSamples);
  if (frames < 0)
    frames = snd_pcm_recover(handle, frames, 0);
  if (frames < 0) {
    printf("snd_pcm_writei failed: %s\n", snd_strerror(err));
  }
  snd_pcm_close(handle);
  return 0;
}
```

Program 5.2 Reading raw data in a C++ program, ALSA library

5.3.3 Reading and Writing Formatted Audio Files in C++

So far, we've worked only with raw audio data. However, most sound is handled in file types that store not only the raw data but also a header containing information about how the data is formatted. Two of the commonly-used file formats are WAV and AIFF. If you want to read or

write a WAV file without the help of an existing library of functions, you need to know the required format.

In WAV and AIFF, data are grouped in **chunks**. Let's look at the WAV format as an example. The format chunk always comes first, containing information about the encoding format, number of channels, sampling rate, data transfer rate, and byte alignment. The samples are stored in the data chunk. The way in which the data is stored depends on whether the file is compressed or not and the bit depth of the file. Optional chunks include the fact, cue, and playlist chunks. To see this format firsthand, you could try taking a RAW audio file, converting it to WAV or AIFF, and then see if it plays in an audio player.

More often, however, it is more to-the-point to use a higher level library that allows you to read and write WAV and AIFF audio files. The *libsndfile* library serves this purpose. If you install this library, you can read WAV or AIFF files into your C++ programs, manipulate them, and write them back to permanent storage as WAV or AIFF files without needing to know the detail of endian-ness or file formats. The website for *libsndfile* has example programs for you to follow. Program 5.3 demonstrates the library with a simple program that opens a sound file (e.g., WAV), reads the contents, reduces the amplitude by half, and writes the result back in the same format.

```
//Give the input and output file names on the command line
#include <stdio.h>
#include <stdlib.h>
#include <string.h>
#include <math.h>
#include <sndfile.h>
#include <iostream>
using namespace std;

#define ARRAY_LEN(x)    ((int) (sizeof (x) / sizeof (x [0])))
#define MAX(x,y)        ((x) > (y) ? (x) : (y))
#define MIN(x,y)        ((x) < (y) ? (x) : (y))

void reduceAmplitude (SNDFILE *, SNDFILE *) ;
sf_count_t sfx_read_double (SNDFILE *, double *, sf_count_t);

int main (int argc, char ** argv) {
  SNDFILE *infile, *outfile ;
  SF_INFO sfinfo ;
  double buffer [1024] ;
  sf_count_t count ;

  if (argc != 3) {
    printf("\nUsage :\n\n    <executable name>  <input file> <output
file>\n") ;
    exit(0);
  }

  memset (&sfinfo, 0, sizeof (sfinfo)) ;
  if ((infile = sf_open (argv [1], SFM_READ, &sfinfo)) == NULL)   {
    printf ("Error : Not able to open input file '%s'\n", argv [1]);
    sf_close (infile);
    exit (1) ;
  }

  if ((outfile = sf_open (argv [2], SFM_WRITE, &sfinfo)) == NULL) {
    printf ("Error : Not able to open output file '%s'\n", argv [argc - 1]);
    sf_close (infile);
    exit (1);
  }

  while ((count = sf_read_double (infile, buffer, ARRAY_LEN (buffer))) > 0) {
    for (int i = 0; i < 1024; i++)
      buffer[i] *= 0.5;
    sf_write_double (outfile, buffer, count);
  }

  sf_close (infile) ;
  sf_close (outfile) ;
  return 0 ;
}
```

Program 5.3 Reading a formatted sound file using the *libsndfile* library

5.3.4 Mathematics and Algorithms for Aliasing

Many of the concepts introduced in Sections 5.1 and 5.2 may become clearer if you experiment with the mathematics, visualize or listen to the audio data in MATLAB, or write some of the algorithms to see the results first-hand. We'll try that in this section.

It's possible to predict the frequency of an aliased wave in cases where aliasing occurs. The algorithm for this is given in Algorithm 5.1. There are four cases to consider. We'll use a sampling rate of 1000 Hz to illustrate the algorithm. At this sampling rate, the Nyquist frequency is 500 Hz.

```
algorithm aliasing {
/*Input:
     Frequency of a single-frequency sound wave to be sampled, f_act
     Sampling rate, f_samp
Output:
     Frequency of the digitized audio wave, f_obs*/
   f_nf = ½ * f_samp   //Nyquist frequency is ½ sampling rate
   if (f_act <= f_nf)                  //Case 1, no aliasing
      f_obs = f_act
   else if (f_nf < f_act <= f_samp)    //Case 2
      f_obs = f_samp - f_act;
   else {
      p = f_act / f_nf   //integer division, drop remainder
      r = f_act mod f_nf      //remainder from integer division
         if (p is even) then           //Case 3
         f_obs = r;
      else if (p is odd) then          //Case 4
         f_obs = f_nf - r;
   }
}
```

Algorithm 5.1

In the first case, when the actual frequency is less than or equal to the Nyquist frequency, the sampling rate is sufficient and there is no aliasing. In the second case, when the actual frequency is between the Nyquist frequency and the sampling rate, the observed frequency is equal to the sampling rate minus the actual frequency. For example, a sampling rate of 1000 Hz causes an 880 Hz sound wave to alias to 120 Hz (Figure 5.35).

Flash Tutorial: Nyquist and Aliasing

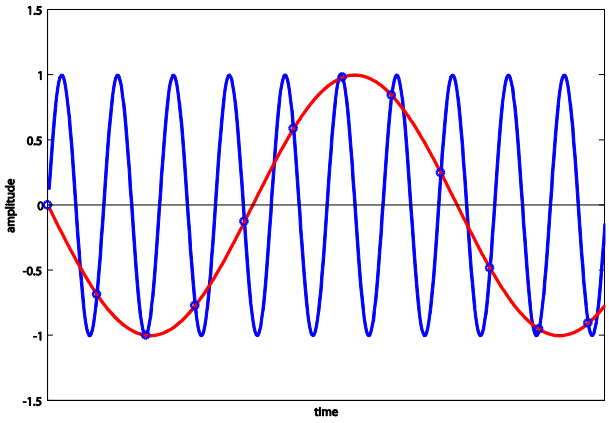


Figure 5.35 880 kHz wave sampled at 1000 kHz

For an actual frequency that is above the sampling rate, like 1320 Hz, the calculations are as follows:

$p = 1320/500 = 2$
$r = 1320 \; mod \; 500 = 320$
p is even, so
$f_{obs} = 320$

Thus, a 1320 Hz wave sampled at 1000 Hz aliases to 320 Hz (Figure 5.36).

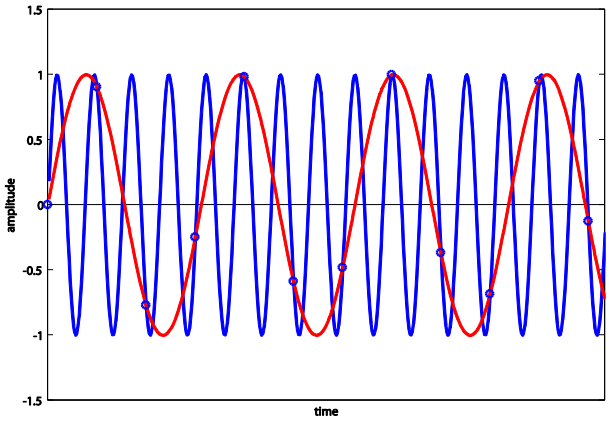


Figure 5.36 1320 kHz wave sampled at 1000 kHz

For an actual frequency of 1760 Hz, we have

$p = 1760/500 = 3$
$r = 1760 \; mod \; 500 = 260$
p is odd, so

$f_{obs} = 500 - 260 = 240$

Thus, a 1760 Hz wave sampled at 1000 Hz aliases to 240 Hz (Figure 5.37).

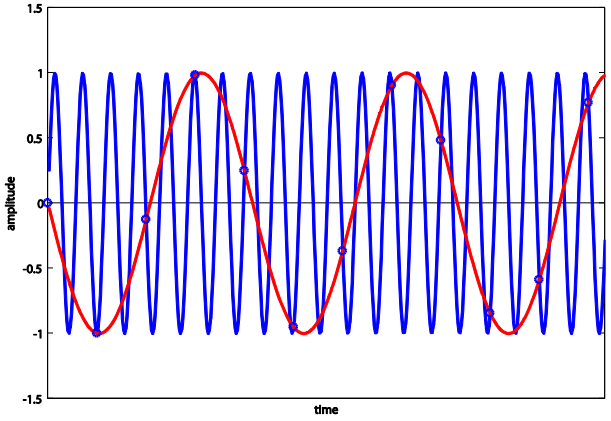


Figure 5.37 1760 kHz wave sampled at 1000 kHz

The graph in Figure 5.38 shows *f_obs* as a function of *f_act* for a fixed sampling rate, in this case 1000 Hz. For a different sampling rate, the graph would still have the same basic repeated-peak shape, but its peaks would be at different places. The first peak from the left is located at the Nyquist frequency on the horizontal axis, which is half the sampling rate. After the first peak, Case 3 is always on the left side of a peak and Case 4 is always on the right.

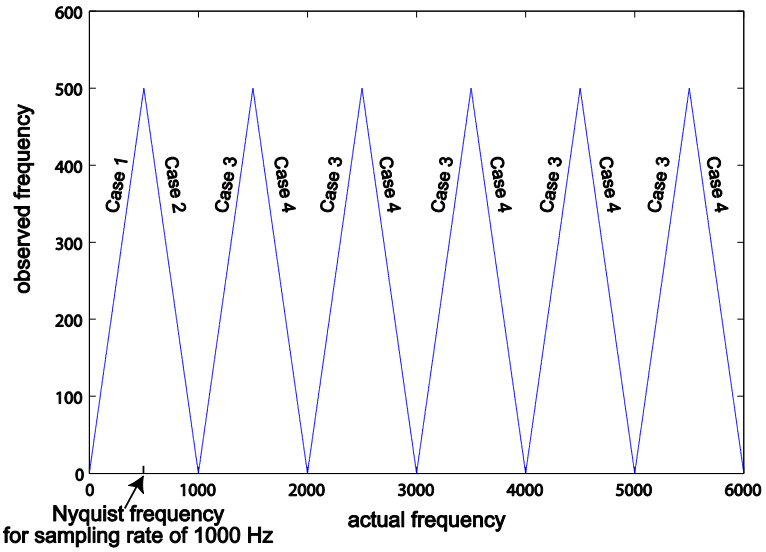


Figure 5.38 Observed frequency plotted against actual frequency for a fixed sampling rate of 1000 Hz

5.3.5 Simulating Sampling and Quantization in MATLAB

Now that you've looked more closely at the process of sampling and quantization in this chapter, you should have a clearer understanding of the MATLAB and C++ examples in Chapters 2 and 3.

In MATLAB, you can generate samples from a sine wave of frequency *f* at a sampling rate *r* for *s* seconds in the following way:

```
f = 440;
sr = 44100;
s = 1;
t = linspace(0,s,sr * s);
y = sin(2*pi*f*t);
```

Aside: An alternative way to get the points at which samples are taken is this:

```
f = 440;
sr = 44100;
s = 1;
t = [1:sr*s];
y = sin(2*pi*f*(t/sr*s));
```

Plugging in the numbers from this example, you get

```
y = sin(2*pi*440*t/44100);
```

We've looked at statements like these in Chapter 2, but let's review. The statement *linspace(0, s, sr * s)* creates a one-dimensional array (which can also be called a vector) of *sr*s* values evenly spaced between 0 and s. These are the points at which the samples are to be taken. One statement in MATLAB can cause an operation to be done on every element of a vector. For example, *y = sin(2*pi*f*t)* takes the sine on each element of *t* and stores the result in vector *y*. Since *t* has 44100 values in it, *y* does also. In this way, MATLAB simulates the sampling process for a single-frequency sound wave.

Quantization can also be simulated in MATLAB. Notice that from the above sequence of commands, all the elements of *y* are between –1 and 1. To quantize these values to a bit depth of *b*, you can do the following:

MATLAB Exercise: Experimenting with Quantization Error

```
b = 8;
sample_max = 2^(b-1)-1;
y_quantized = floor(y*sample_max);
```

The plot function graphs the result.

```
plot(t, y);
```

If we want to zoom in on the first, say, 500 values, we can do so with

```
plot(t(1:500), y(1:500));
```

With these basic commands, you can do the suggested exercise linked with this section, which has you experiment with quantization error and dynamic range at various bit depths.

5.3.6 Simulating Sampling and Quantization in C++

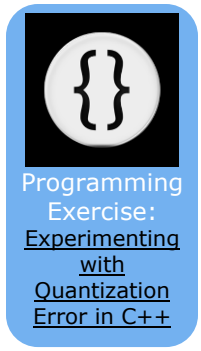
You can simulate sampling and quantization in C++ just as you can in MATLAB. The difference is that you need loops to operate on arrays in C++, while one command in MATLAB can cause an operation to be performed on each element in an array. (You can also write loops in MATLAB, but it isn't necessary where array operations are available.) For example, the equivalent of the MATLAB lines above is given in C++ below. (We use mostly C syntax but,

for simplicity, include some C++ features like dynamic allocation with *new* and the ability to declare variables anywhere in the program.)

```
int f = 440;
int r = 44100;
double s = 1;
int b = 8;
double* y = new double[r*s];
for (int t = 0; t < r * s; t++)
      y[t] = sin(2*PI*f*(t/(r*s)));
```

To quantize the samples in *y* to a bit depth of *b*, we do this:

```
int sample_max = pow(2, b-1) - 1;
short* y_quantized = new short[r*s];
for (int i = 0; i < r * s; i++)
      y_quantized[i] = floor(y * sample_max);
```



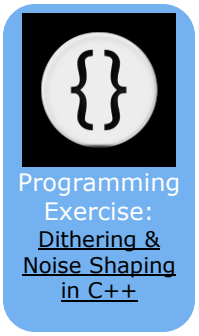
The suggested programming exercise linked with this section uses code such as this to implement Algorithm 5.1 for aliasing and to experiment with quantization error at various bit depths.

5.3.7 The Mathematics of Dithering and Noise Shaping

Dithering is described in Section 5.1.2.5 as the addition of low-amplitude random noise to an audio signal as it is being quantized. The purpose of dithering is to prevent neighboring sample values from quantizing all to the same level, which can cause breaks or choppiness in the sound. Noise shaping can be performed in conjunction with dithering to raise the noise to a higher frequency where it is not noticed as much. Now let's look at the mathematics of dithering and noise shaping.

First, the amount of random noise to be added to each sample must be determined. A probability density function can be used for this purpose. We'll use the **triangular probability density function** graphed in Figure 5.39 as an example. This graph depicts the following:

- There is 0 probability that an amount less than –1 and greater than 1 will be added to any given sample.
- As *x* moves from –1 to 0 and as *x* moves from 1 to 0, there is an increasing probability that *x* will be added to any given sample.



Thus, it's most likely that some number close to 0 will be added as dither, and the highest magnitude value to be added or subtracted is 1. If you were to implement dithering yourself in MATLAB or a C++ program (as in the exercises associated with this section), you could create a triangular probability density function by getting a random number between –1 and 0 and another between 0 and 1 and summing them.

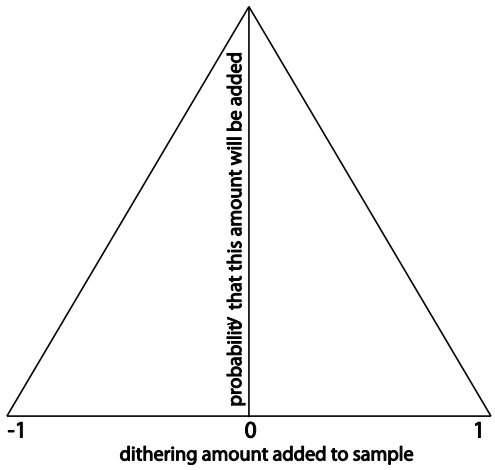


Figure 5.39 Triangular probability density function

Audio processing programs like Adobe Audition or Sound Forge offer a variety of probability functions for noise shaping, as shown in Figure 5.40. The rectangular probability density function gives equal probability for all numbers within a given range. The Gaussian function weighs the probabilities according to a Gaussian rather than a triangular shape, so it would look like Figure 5.39 except more bell-shaped. This creates noise that is more like common environmental noise, like tape hiss.

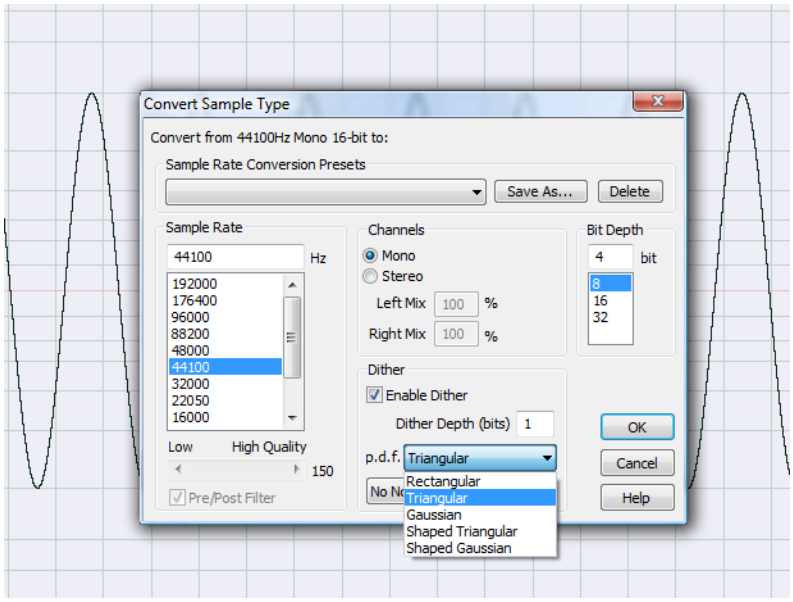


Figure 5.40 Probability functions for noise shaping

Requantization without dithering, with dithering, and with dithering and noise shaping are compared in Figure 5.41. A 16-bit audio file has been requantized to 4 bits (an unlikely scenario, but it makes the point). From these graphs, it may look like noise shaping adds additional noise. In fact, the reason the noise graph looks more dense when noise shaping is added is that the frequency components are higher.

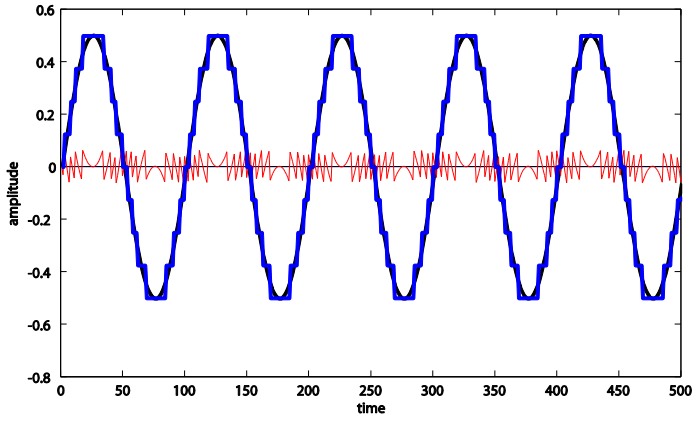
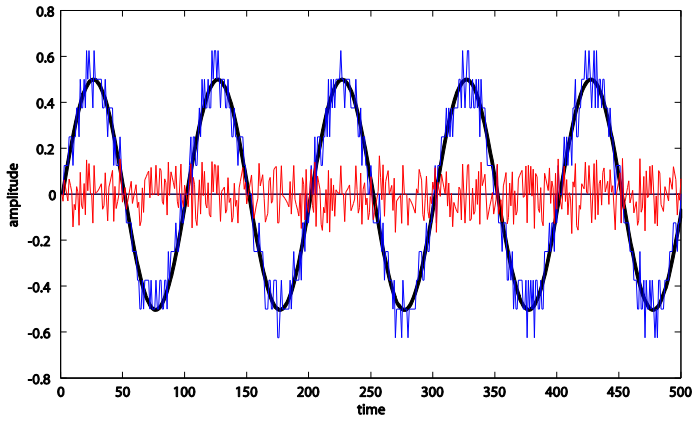
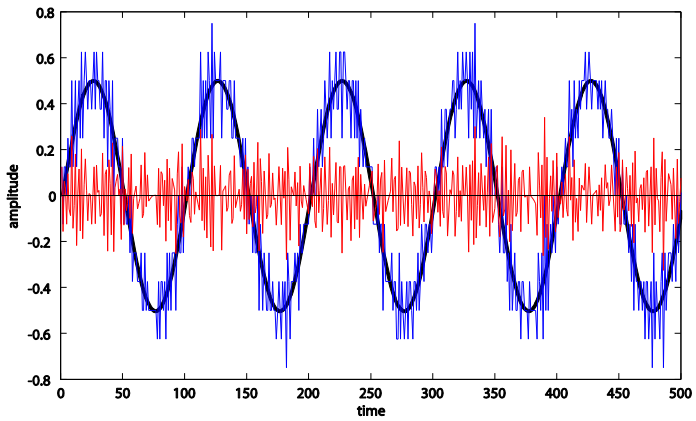
	original 16-bit wave, wave requantized at 4 bits, and error wave
	original 16-bit wave, wave requantized at 4 bits, and error wave with dithering
	original 16-bit wave, wave requantized at 4 bits, and error wave with dithering and noise shaping

Figure 5.41 Comparison of requantization without dithering, with dithering, and with dithering and noise shaping

To examine this more closely, we can extract the noise into a separate audio file by subtracting the original file from the requantized one. We've done this for both cases – noise from dither, and noise from dither and noise shaping. (The noise includes the quantization error.) Closeups of the noise graphs are shown in Figure 5.42. What you should notice is that dither-with-noise-shaping noise has higher frequency components than dither-only noise.

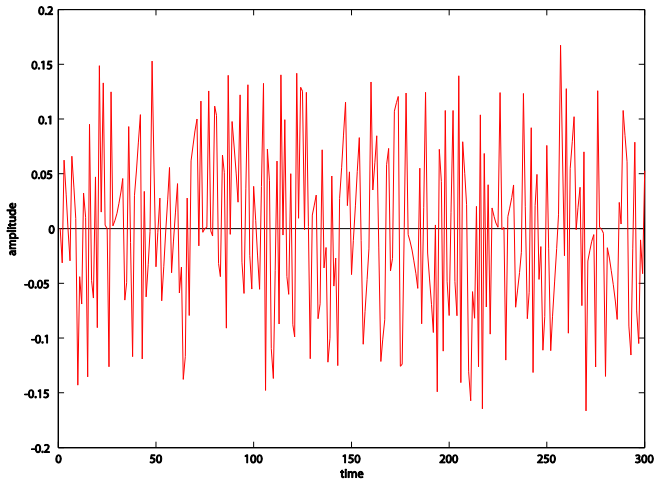
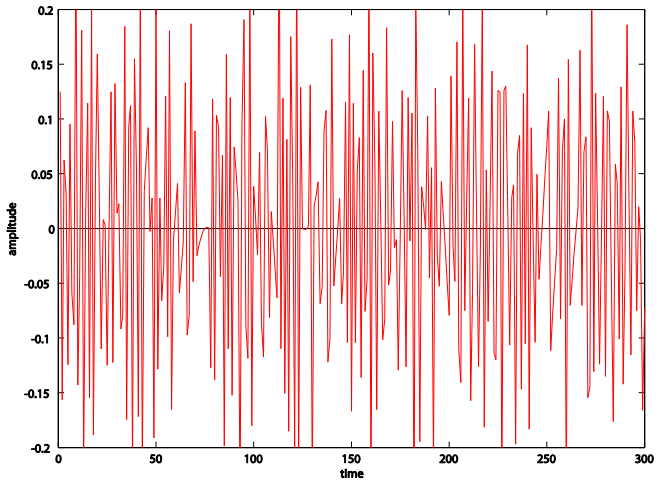
	noise from dither only
	noise from dither with noise shaping

Figure 5.42 Closeup of noise from dither and noise from dither with noise shaping (quantization noise included)

You can see this also in the spectral view of the noise files in Figure 5.43. In a **spectral view of an audio file**, time is on the *x*-axis, frequency is on the *y*-axis, and the amplitude of the frequency is represented by the color at each (*x,y*) point. The lowest amplitude is represented by blue, medium amplitudes move from red to orange, and the highest amplitudes move from yellow to white. You can see that when dithering alone is applied, the noise is spread out over all frequencies. When noise shaping is added to dithering, there is less noise at low frequency and more noise at high frequency. The effect is to pull more of the noise to high frequencies, where it is noticed less by human ears. (ADCs also can filter out the high frequency noise if it is above the Nyquist frequency.)

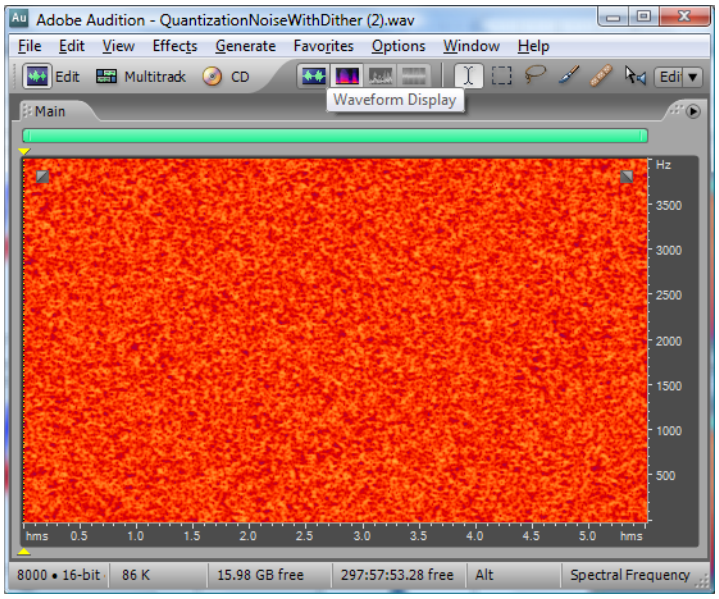
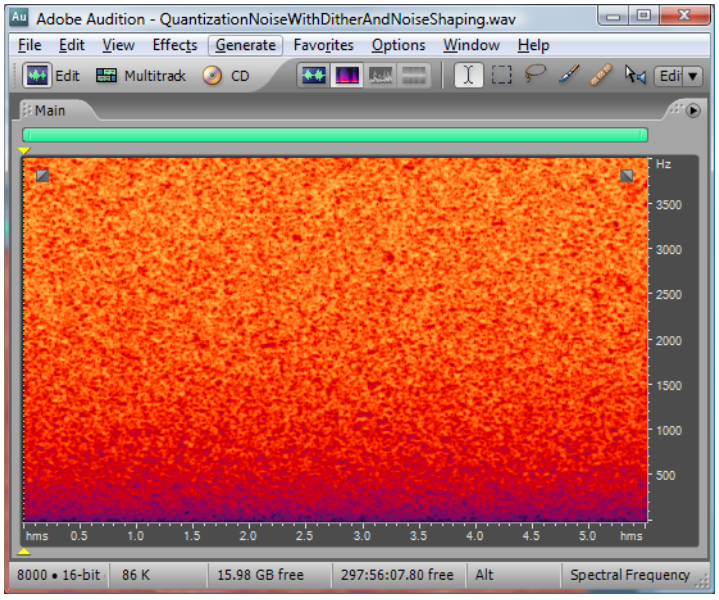
	noise from dither only
	noise from dither and noise shaping

Figure 5.43 Spectral view of noise from dithering and noise from dithering with noise shaping (quantization noise included)

Noise shaping algorithms, first developed by Cutler in the 1950s, operate by computing the error from quantizing a sample (including the error from dithering and noise shaping) and adding this error to the next sample before it is quantized. If the error from the i^{th} sample is positive, then, by subtracting the error from the next sample, noise shaping makes it more likely that the error for the $i+1^{st}$ sample will be negative. This causes the error wave to go up and down more frequently; i.e., the frequency of the error wave is increased. The algorithm is given

in Algorithm 5.2. You can output your files as uncompressed RAW files, import the data into MATLAB, and graph the error waves, comparing the shape of the error with dither-only and with noise shaping using various values for *c*, the scaling factor. The algorithm given is the most basic kind of noise shaping you can do. Many refinements and variations of this algorithm have been implemented and distributed commercially.

```
algorithm noise_shape {
/*Input:
     b_orig, the original bit depth
     b_new, the new bit depth to which samples are to be quantized
     F_in, an array of N digital audio samples that are to be
     quantized, dithered, and noise shaped.  It's assumed that these are read
     in from a RAW file and are values between
     -2^b_orig-1 and (2^b_orig-1)-1.
     c, a scaling factor for the noise shaping
Output:
     F_out, an array of N digital audio samples quantized to bit
     depth b_new using dither and noise shaping*/

   s = (2^b_orig)/(2^b_new);
   c = 0.8;  //Other scaling factors can be tried.*/
   e = 0;
   for (i = 0; i < N; i++) {
/*Get a random number between –1 and 1 from some probability density
function*/
      d = pdf();
      F_scaled  = F_in[i] / s;  //Integer division, discarding remainder
      F_scaled_plus_dith_and_error = F_scaled + d + c*e;
      F_out[i] =  floor(F_scaled_plus_dith_and_error);
      e = F_scaled – F_out[i];
   }
}
```

Algorithm 5.2

5.3.8 Algorithms for Audio Companding and Compression

5.3.8.1 Mu-law Encoding

Mu-law encoding (also denoted ***μ*-law)** is a nonlinear companding method used in telecommunications. **Companding** is a method of compressing a digital signal by reducing the bit depth before it is transmitted and then expanding it when it is received. The advantage of mu-law encoding is that it preserves some of the dynamic range that would be lost if a linear method of reducing the bit depth were used instead. Let's look more closely at the difference between linear and nonlinear methods of bit depth reduction.

Picture this scenario. A telephone audio signal is to be transmitted. It is originally 16 bits. To reduce the amount of data that has to be transmitted, the audio data is reduced to 8 bits. In a linear method of bit depth reduction, the amount of quantization error possible at low amplitudes is the same as the amount of error possible at high amplitudes. However, this means that the *ratio* of the error to the sample value is *greater* for low amplitude samples than for high. Let's assume that in converting from 16 bits to 8 bits is done by dividing by the sample value by 256 and rounding down. Consider a 16-bit sample that is originally 32767, the highest positive

value possible. (16-bit samples range from –32768 to 32767.) Converted to 8 bits, the sample value becomes 127. (32767/256 = 127 with a remainder of 255.) The error from rounding is 255/32768. This is an error of less than 1%. But compare this to the error for the lowest magnitude 16-bit samples, those between 0 and 255. They all round to 0 when reduced to a bit depth of 8, which is 100% error. The point is that with a linear bit depth reduction method, rounding down has a greater impact on low amplitude samples than on high amplitude ones.

In a nonlinear method such as mu-law encoding, when the bit depth is reduced, the effect is that not all samples are encoded in the same number of bits. Low amplitude samples are given more bits to protect them from quantization error.

Equation 5.5 (which is, more precisely, a function) effectively redistributes the sample values so that there are more quantization levels at lower amplitudes and fewer quantization levels at higher amplitudes. For requantization from 16 to 8 bits, $\mu = 255$.

Let x be a sample value, $-1 \leq x \leq 1$. Let $sign(x) = -1$ if x is negative and $sign(x) = 1$ otherwise. Then the **mu-law function** (also denoted μ**-law**) is defined by

$$mu(x) = sign(x)\left(\frac{\ln(1+\mu|x|)}{\ln(1+\mu)}\right) = sign(x)\left(\frac{\ln(1+255|x|)}{5.5452}\right) for\ \mu = 255$$

Equation 5.5

The graph of the mu-law function in Figure 5.44 has the original sample value (on a scale of –1 to 1) on the x-axis and the sample value after it is run through the mu-law function on the y-axis. Notice that the difference between, say, *mu(0.001)* and *mu(0.002)* is greater than the difference between *mu(0.981)* and *mu(0.982)*.

mu(0.001) = 0.0410
mu(0.002) = 0.0743
mu(0.981) = 0.9966
mu(0.982) = 0.9967

The mu-law function has the effect of "spreading out" the quantization intervals more at lower amplitudes.

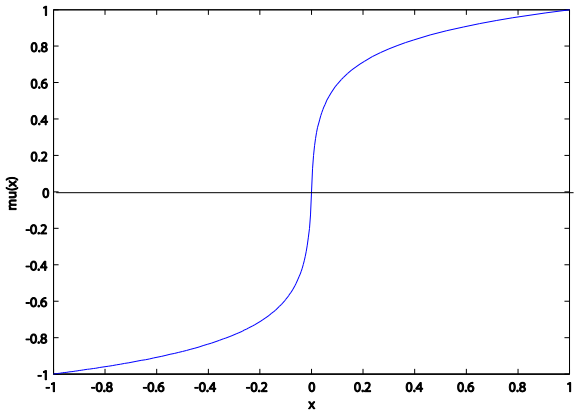


Figure 5.44 Graph of μ-law function

It may be easier to see how this works if we start with sample values on a 16-bit scale (i.e., between –32768 and 32767) and then scale the results of the mu-law function back to 8 bits. For example, what happens if we begin with sample values of 33, 66, 32145, and 32178? (Notice that these normalize to 0.001, 0.002, 0.981, and 0.982, respectively, on a –1 to 1 scale.)

mu(33/32768) = 0.0412;
*floor(128*0.0412) = 5;*
mu(66/32768) = 0.0747;
floor(128 0.0747) = 9;*
mu(32145/32768) = 0.9966;
*floor(128*0.9966) = 127;*
mu(32178/32768) = 0.9967;
*floor(128*0.9967) = 127;*

The sample values 33 and 66 on a 16-bit scale become 5 and 9, respectively, on an 8-bit scale. The sample values 32145 and 32178 both become 127 on an 8-bit scale. Even though the difference between 66 and 33 is the same as the difference between 32178 and 32145, 66 and 33 fall to different quantization levels when they are converted to 8 bits, but 32178 and 32145 fall to the same quantization level. There are more quantization levels at lower amplitudes after the mu-law function is applied.

The result is that there is less error at low amplitudes, after requantization to 16 bits, than there would have been if a linear quantization method had been used. Equation 5.6 shows the function for expanding back to the original bit depth.

Let x be a μ-law encoded sample value, $-1 \leq x \leq 1$. Let $sign(x) = -1$ if x is negative and $sign(x) = 1$ otherwise. Then the **inverse μ-law function** is defined by

$$\begin{pmatrix} mu\_inverse(x) = sign(x)\left(\frac{(\mu+1)^{|x|}-1}{\mu}\right) = \\ sign(x)\left(\frac{256^{|x|}-1}{255}\right) for\ \mu = 255 \end{pmatrix}$$

Equation 5.6

Our original 16-bit values of 33 and 66 convert to 5 and 9, respectively, in 8 bits. We can convert them back to 16 bits using the inverse mu-law function as follows:

mu_inverse(5/128) = .00094846
ceil(32768 .00094846) = 32*
mu_inverse(9/128) = 0.0019
*ceil(32768*0.0019) = 63*

The original value of 33 converted back to 32. The original value of 66 converted back to 63. On the other hand, look what happens to the higher amplitude values that were originally 32145 and 32178 and converted to 127 at 8 bits.

mu_inverse(127/128) = 0.9574
*ceil(32768*0.9574) = 31373*

Both 32145 and 32178 convert back to 31373.

With mu-law encoding, the results are better than they would have been if a linear method of bit reduction had been used. Let's look at the linear method. Again suppose we convert from 16 to 8 bits by dividing by 256 and rounding down. Then both 16-bit values of 33 and 66 would convert to 0 at 8 bits. To convert back up to 16 bits, we multiply by 256 again and still have 0. We've lost all the information in these samples. On the other hand, by a linear method both 32145 and 32178 convert to 125 at 8 bits. When they're scaled back to 16 bits, they both become 32000. A comparison of percentage errors for all of these samples using the two conversion methods is shown in Table 5.2. You can see that mu-law encoding preserves information at low amplitudes that would have been lost by a linear method. The overall result is that with a bit depth of only 8 bits for transmitting the data, it is possible to retrieve a dynamic range of about 12 bits or 72 dB when the data is uncompressed to 16 bits (as opposed to the 48 dB expected from just 8 bits). The dynamic range is increased because fewer of the low amplitude samples fall below the noise floor.

original sample at 16 bits	**sample after linear companding, 16 to 8 to 16**	**error**	**sample after nonlinear companding, 16 to 8 to 16**	**percentage error**
33	0	100%	32	3%
66	0	100%	63	4.5%
32145	32000	0.45%	31373	2.4%
32178	32000	0.55%	31373	2.5%

Table 5.2 Comparison of error in linear and nonlinear companding

Rather than applying the mu-law function directly, common implementations of mu-law encoding achieve an equivalent effect by dividing the 8-bit encoded sample into a sign bit, mantissa, and exponent, similar to the way floating point numbers are represented. An advantage of this implementation is that it can be performed with fast bit-shifting operations. The programming exercise associated with this section gives more information about this implementation, comparing it with a literal implementation of the mu-law and inverse mu-law functions based on logarithms.

Mu-law encoding (and its relative, A-law, which is used Europe) reduces the amount of data in an audio signal by quantizing low-amplitude signals with more precision than high amplitude ones. However, since this bit-depth reduction happens in conjunction with digitization, it might more properly be considered a conversion rather than a compression method.

Actual audio compression relies on an analysis of the frequency components of the audio signal coupled with a knowledge of how the human auditory system perceives sound – a field of study called **psychoacoustics**. Compression algorithms based on psychoacoustical models are grouped under the general category of **perceptual encoding**, as is explained in the following section.

5.3.8.2 Psychoacoustics and Perceptual Encoding

Psychoacoustics is the study of how the human ears and brain perceive sound. Psychoacoustical experiments have shown that human hearing is nonlinear in a number of ways, including perception of octaves, perception of loudness, and frequency resolution.

For a note C2 that is one octave higher than a note C1, the frequency of C2 is twice the frequency of C1. This implies that the frequency width of a lower octave is smaller than the frequency width of a higher octave. For example, the octave C5 to C6 is twice the width of C4 to C5. However, the octaves are perceived as being the same width in human hearing. In this sense, our perception of octaves is nonlinear.

Humans hear best (i.e., have the most sensitivity to amplitude) in the range of about 1000 to 5000 Hz, which is close to the range of the human voice. We hear less well at both ends of the frequency spectrum. This is illustrated in Figure 5.45, which shows the shape of the threshold of hearing plotted across frequencies.

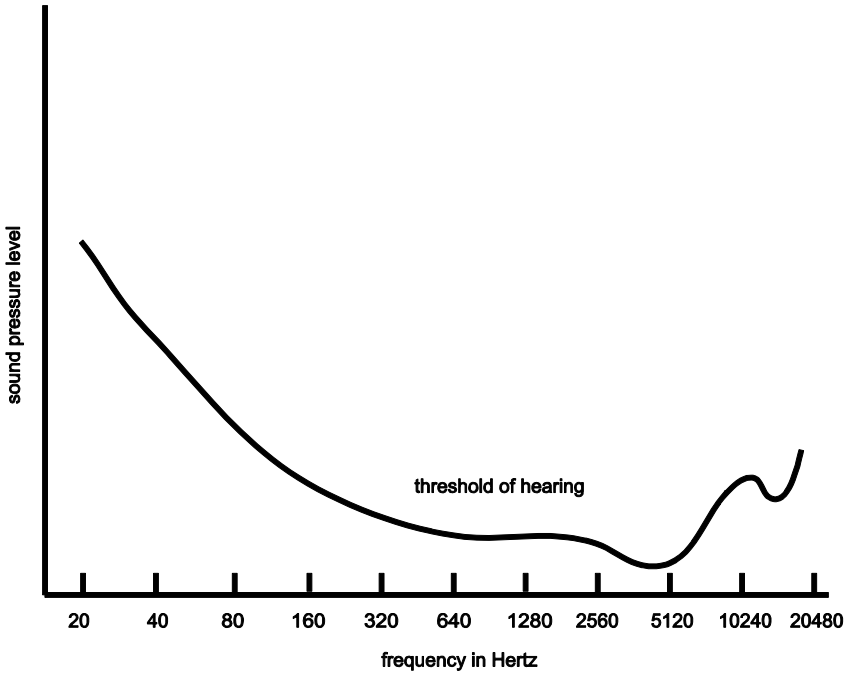


Figure 5.45 Threshold of hearing plotted over frequencies in range of human hearing

Human ability to distinguish between frequencies decreases nonlinearly from low to high frequencies. At the very lowest audible frequencies, we can tell the difference between pitches that are only a few Hz apart, while at high frequencies the pitches must be separated by more than 100 Hz before we notice a difference. This difference in frequency sensitivity arises from the fact that the inner ear is divided into **critical bands**. Each band is tuned to a range of frequencies. Critical bands for low frequencies are narrower than those for high ones. Between frequencies of 1 and 500 Hz, critical bands have a width of about 100 Hz or less, whereas the width of the critical band at the highest audible frequency is about 4000 Hz.

The goal of applying psychoacoustics to compression methods is to determine the components of sounds that human ears don't perceive very well, if at all. These are the parts that can be discarded, thereby decreasing the amount of data that must be stored in digitized sound. An understanding of critical bands within the human ear helps in this regard. Within a band, a

phenomenon called **masking** can occur. Masking results when two frequencies are received by a critical band at about the same moment in time, one of the frequencies being significantly louder than the first such that it makes the first inaudible. The loud frequency is called the **masking tone**, and the quiet one is the **masked frequency**. Masking causes the threshold of hearing to be raised within a critical band in the presence of a masking tone. The new threshold of hearing is called the **masking threshold**, as depicted in Figure 5.46. Note that this graph represents the masking tone in one critical band in a narrow window of time. Audio compression algorithms look for places where masking occurs in all of the critical bands over each consecutive window of time for the duration of the audio signal.

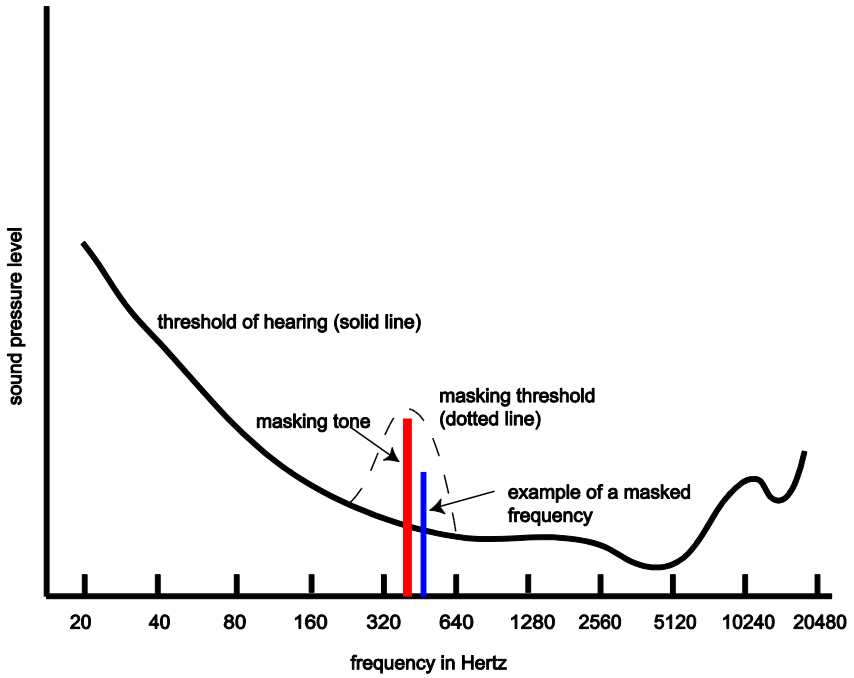


Figure 5.46 Threshold of hearing altered in presence of masking tone

Another type of masking, called **temporal masking**, occurs with regard to transients. **Transients** are sudden bursts of sounds like drum beats, hand claps, finger snaps, and consonant sounds. When two transients occur close together in time, the louder one can mask the weaker one. Identifying instances of temporal masking is another important part of perceptual encoding.

5.3.8.3 MP3 and AAC Compression

Two of the best known compression methods that use perceptual encoding are MP3 and AAC. MP3 is actually a shortened name for MPEG-1 Audio Layer III. It was developed through a collaboration of scientists at Fraunhofer IIS, AT&T Bell Labs, Thomason-Brandt, CCETT, and others and was approved by ISO/IEC as part of the MPEG standard in 1991. MP3 was a landmark compression method in the way that it popularized the exchange of music by means of the web. Although MP3 is a lossy compression

Aside: MPEG is an acronym for Motion Picture Experts Group and refers to a family of compression methods for digital audio and video. MPEG standards were developed in phases and layers beginning in 1988, as outlined in Table 5.3. The phases are indicated by Arabic numerals and the layers by Roman numerals.

method in that it discards information that cannot be retrieved, music lovers were quick to accept the tradeoff between some loss of audio quality and the ability to store and exchange large numbers of music files.

AAC (Advanced Audio Coding) is the successor of MP3 compression, using similar perceptual encoding techniques but improving on these and supporting more channels, sampling rates, and bit rates. AAC is available within both the MPEG-2 and MPEG-4 standards and is the default audio compression method for iPhones, iPods, iPads, Android phones, and a variety of video game consoles. Table 5.1 shows the filename extensions associated with MP3 and AAC compression.

You may have read that MP3 offers a compression ratio of about 11 to 1. This number gives you a general idea, but exact compression ratios vary. Table 5.3 shows ranges of bit rates available for different layers within the MPEG standard. Typically, an MP3 compressor offers a number of choices of bit rates, these varying with sampling rates, as shown in Figure 5.47 for Adobe Audition's MP3/mp3Pro encoder. The bit rate indicates how many bits of compressed data is generated per second. A constant bit rate of 128 kb/s for mono audio sampled at 44100 Hz with 16 bits per sample gives a compression ratio of 5.5:1, as shown in the calculation below.

$$44100\ samples/s * 16 = 705600\ b/s\ uncompressed$$
$$705600\ uncompressed : 128000\ b/s\ compressed\ \approx 5.5:1$$

For stereo (32 bits per sample) at a sampling rate of 44100, compressed to 128 kb/s, the compression ratio is approximately 11:1. This is where the commonly-cited compression ratio comes from – the fact that frequently, MP3 compression is used on CD quality stereo at a bit rate of 128 kb/s, which most listeners seem to think gives acceptable quality. For a given implementation of MP3 compression and a fixed sampling rate, bit depth, and number of channels, a higher bit rate yields better quality in the compressed audio than a lower bit rate. When fewer bits are produced per second of audio, more information must be thrown out.

Version	Bit rates*	Applications	Channels	Sampling rates supported
MPEG-1		CD, DAT, ISDN, video games, digital audio broadcasting, music shared on the web, portable music players	mono or stereo	32, 44.1, and 48 kHz (and 16, 22.05, and 24 kHz for MPEG-2 LSF, low sampling frequency)
Layer I	32−448 kb/s			
Layer II	32−384 kb/s			
Layer III	32−320 kb/s			
MPEG-2		multichannels, multilingual extensions	5.1 surround	32, 44.1, and 48 kHz
Layer I	32−448 kb/s			
Layer II	32−384 kb/s			
Layer III	32−320 kb/s			
AAC in MPEG-2 and MPEG-4 (Advanced Audio Coding)	8 −384 kb/s	multichannels, music shared on the web, portable music players, cell phones	up to 48 channels	32, 44.1, and 48 kHz and other rates between 8 and 96 kHz
*Available bit rates vary with sampling rate.				

Table 5.3 MPEG audio

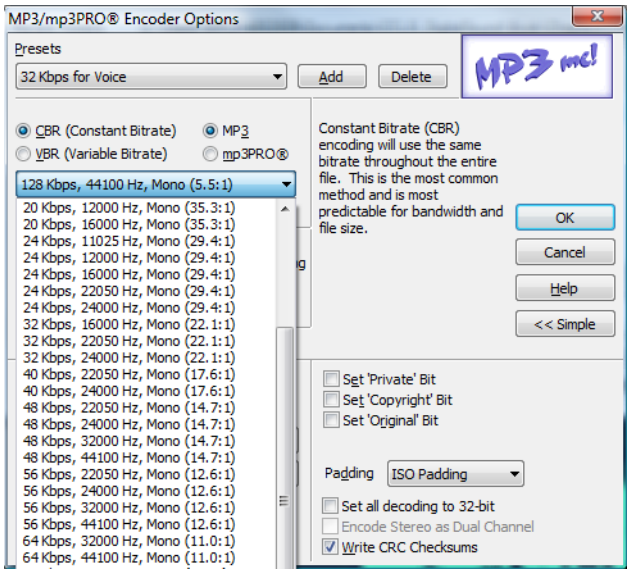


Figure 5.47 Bit rate choices in Adobe Audition's MP3/mp3PRO encoder

The MPEG standard does not prescribe how compression is to be implemented. Instead, it prescribes the format of the encoded audio signal, which is a bit stream made up of frames with certain headers and data fields. The MPEG standard suggests two psychoacoustical models that could accomplish the encoding, the second model, used for MP3, being more complex and generally resulting in greater compression than the first. In reality, there can be great variability in the time efficiency and quality of MP3 encoders. The advantage to not dictating details of the compression algorithm is that implementers can compete with novel implementations and refinements. If the format of the bit stream is always the same, including what each part of the bit stream represents, then decoders can be implemented without having to know the details of the encoder's implementation.

An MP3 bit stream is divided into frames each of which represents the compression of 1152 samples. The size of the frame containing the compressed data is determined by the bit rate, a user option that is set prior to compression. As before, let's assume that a bit rate of 128 kb/s is chosen to compress an audio signal with a sampling rate of 44.1 kHz. This implies that the number of frames of compressed data that must be produced per second is $44100/1152 \approx 38.28$. The chosen bit rate of 128 kb/s is equal to 16000 bytes per second. Thus, the number of bytes allowed for each frame is $16000/38.28 \approx 418$. You can see that if you vary the sampling rate or the bit rate, there will be a different number of bytes in each frame. This computation considers only the data part of each compressed frame. Actually, each frame consists of a 32-byte header followed by the appropriate number of data bytes, for a total of approximately 450 bytes in this example.

The format of the header is shown in Figure 5.48 and explained in Table 5.4.

1 1 1 1 1 1 1 1 1 1 1	1 1	0 1	1	0 1 0 0	0 0	0	0	0 1	0 0	0	0	0 0
11 bit sync	version	layer	error prot.	bit rate	freq.	pad bit	priv. bit	mode	mode ext.	copy	orig.	emphasis
A	B	C	D	E	F	G	H	I	J	K	L	M

Figure 5.48 MP3 frame header

Field	Length in Bits	Contents
A	11	all 1s to indicate start of frame
B	2	MPEG-1 indicated by bits 1 1
C	2	Layer III indicated by bits 0 1
D	1	0 means protected by CRC* (with 16 bit CRC following header) 1 means not protected
E	4	bit rate index 0000 free 0001 32 kb/s 0010 40 kb/s 0011 48 kb/s 0100 56 kb/s 0101 64 kb/s 0110 80 kb/s 0111 96 kb/s 1000 112 kb/s 1001 128 kb/s 1010 160 kb/s 1011 192 kb/s 1100 224 kb/s 1101 256 kb/s 1110 320 kb/s 1111 bad
F	2	frequency index 00 44100 Hz 01 48000 Hz 10 32000 Hz 11 reserved
G	1	0 frame is not padded 1 frame is padded with an extra byte Padding is used so that bit rates are fitted exactly to frames. For example, a 44100 Hz audio signal at 128 kb/s yields a frame of about 417.97 bytes
H	1	private bit, used as desired by implementer
I	2	channel mode 00 stereo 01 joint stereo 10 2 independent mono channels 11 1 mono channel
J	2	mode extension used with joint stereo mode
K	1	0 if not copyrighted 1 if copyrighted
L	1	0 if copy of original media 1 if not
M	2	indicates emphasized frequencies
*CRC stands for cyclical redundancy check, an error correcting code that checks for errors in the transmission of data.		

Table 5.4 Contents of MP3 frame header

MP3 supports both **constant bit rate** (**CBR**) and **variable bit rate** (**VBR**). VBR makes it possible to vary the bit rate frame-by-frame, allocating more bits to frames that require greater

frequency resolution because of the complexity of the sound in that part of the audio. This can be implemented by allowing the user to specify the maximum bit rate. An alternative is to allow the user to specify an average bit rate. Then the bit rate can vary frame-by-frame but is controlled so that it averages to the chosen rate.

A certain amount of variability is possible even within CBR. This is done by means of a **bit reservoir** in frames. A bit reservoir is a created from bits that do not have to be used in a frame because the audio being encoded is relatively simple. The reservoir provides extra space in a subsequent frame that may be more complicated and requires more bits for encoding.

Field I in Table 5.4 makes reference to the **channel mode**. MP3 allows for one mono channel, two independent mono channels, two stereo channels, and **joint stereo mode**. Joint stereo mode takes advantage of the fact that human hearing is less sensitive to the location of sounds that are the lowest and highest ends of the audible frequency ranges. In a stereo audio signal, low or high frequencies can be combined into a single mono channel without much perceptible difference to the listener. The joint stereo option can be specified by the user.

A sketch of the steps in MP3 compression is given in Algorithm 5.3 and the algorithm is diagrammed in Figure 5.49. This algorithm glosses over details that can vary by implementation but gives the basic concepts of the compression method.

```
algorithm MP3 {
/*Input:  An audio signal in the time domain
  Output:  The same audio signal, compressed
/*
Process the audio signal in frames
For each frame {
   Use the Fourier transform to transform the time domain data to the frequency domain, sending
      the results to the psychoacoustical analyzer {
         Based on masking tones and masked frequencies, determine the signal-to-masking noise
            ratios (SMR) in areas across the frequency spectrum
         Analyze the presence and interactions of transients
      }
   Divide the frame into 32 frequency bands
   For each frequency band {
      Use the modified discrete cosine transform (MDCT) to divide each of the 32 frequency bands
         into 18 subbands, for a total of 576 frequency subbands
      Sort the subbands into 22 groups, called scale factor bands, and based on the SMR, determine
         a scaling factor for each scale factor band
      Use nonuniform quantization combined with scaling factors to quantize
      Encode side information
      Use Huffman encoding on the resulting 576 quantized MDCT coefficients
      Put the encoded data into a properly formatted frame in the bit stream
   }
}
```

Algorithm 5.3 MP3 compression

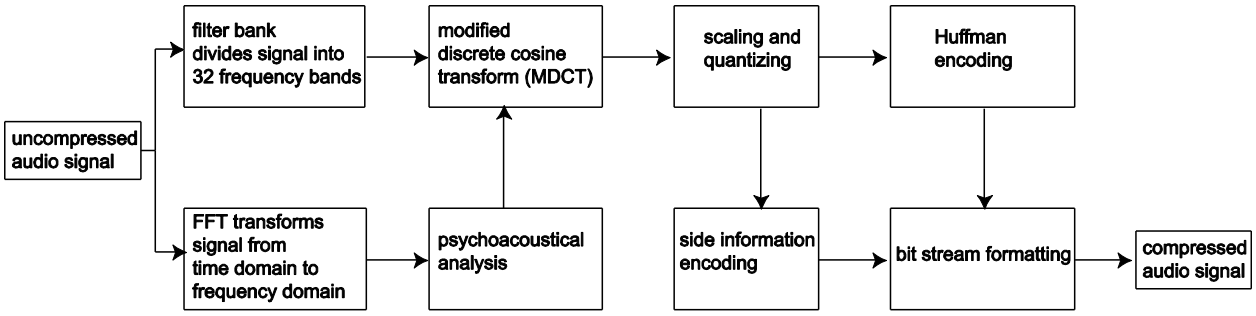


Figure 5.49 MP3 compression

Let's look at the steps in this algorithm more closely.

1. Divide the audio signal in frames.

MP3 compression processes the original audio signal in frames of 1152 samples. Each frame is split into two granules of 576 samples each. Frames are encoded in a number of bytes consistent with the bit rate set for the compression at hand. In the example described above (with sampling rate of 44.1 kHz and requested bit rate of 128 kb/s), 1152 samples are compressed into a frame of approximately 450 bytes – 418 bytes for data and 32 bytes for the header.

2. Use the Fourier transform to transform the time domain data to the frequency domain, sending the results to the psychoacoustical analyzer.

The fast Fourier transform changes the data to the frequency domain. The frequency domain data is then sent to a psychoacoustical analyzer. One purpose of this analysis is to identify masking tones and masked frequencies in a local neighborhood of frequencies over a small window of time. The psychoacoustical analyzer outputs a set of **signal-to-mask ratios** (**SMRs**) that can be used later in quantizing the data. The SMR is the ratio between the amplitude of a masking tone and the amplitude of the minimum masked frequency in the chosen vicinity. The compressor uses these values to choose scaling factors and quantization levels such that quantization error mostly falls below the masking threshold. Step 5 explains this process further.

Another purpose of the psychoacoustical analysis is to identify the presence of transients and temporal masking. When the MDCT is applied in a later step, transients have to be treated in smaller window sizes to achieve better time resolution in the encoding. If not, one transient sound can mask another that occurs close to it in time. Thus, in the presence of transients, windows are made one third their normal size in the MDCT.

3. Divide each frame into 32 frequency bands

Steps 2 and 3 are independent and actually could be done in parallel. Dividing the frame into frequency bands is done with filter banks. Each filter bank is a bandpass filter that allows only a range of frequencies to pass through. (Chapter 7 gives more details on bandpass filters.) The complete range of frequencies that can appear in the original signal is 0 to ½ the sampling rate, as we know from the Nyquist theorem. For example, if the sampling rate of the signal is 44.1 kHz, then the highest frequency that can be present in the signal is 22.05 kHz. Thus, the filter banks yield 32 frequency bands between 0 and 22.05 kHz, each of width 22050/32, or about 689 Hz.

The 32 resulting bands are still in the time domain. Note that dividing the audio signal into frequency bands increases the amount of data by a factor of 32 at this point. That is, there are 32 sets of 1152 time-domain samples, each holding just the frequencies in its band. (You can

understand this better if you imagine that the audio signal is a piece of music that you decompose into 32 frequency bands. After the decomposition, you could play each band separately and hear the musical piece, but only those frequencies in the band. The segments would need to be longer than 1152 samples for you to hear any music, however, since 1152 samples at a sampling rate of 44.1 kHz is only 0.026 seconds of sound.)

4. Use the MDCT to divide each of the 32 frequency bands into 18 subbands for a total of 576 frequency subbands.

The MDCT, like the Fourier transform, can be used to change audio data from the time domain to the frequency domain. Its distinction is that it is applied on overlapping windows in order to minimize the appearance of spurious frequencies that occur because of discontinuities at window boundaries. ("Spurious frequencies" are frequencies that aren't really in the audio, but that are yielded from the transform.) The overlap between successive MDCT windows varies depending on the information that the psychoacoustical analyzer provides about the nature of the audio in the frame and band. If there are transients involved, then the window size is shorter for greater temporal resolution. Otherwise, a larger window is used for greater frequency resolution.

5. Sort the subbands into 22 groups, called scale factor bands, and based on the SMR, determine a scaling factor for each scale factor band. Use nonuniform quantization combined with scaling factors to quantize.

Values are raised to the ¾ power before quantization. This yields nonuniform quantization, aimed at reducing quantization noise for lower amplitude signals, where it has a more harmful impact.

The psychoacoustical analyzer provides information that is the basis for sorting the subbands into **scale factor bands**. The scale factor bands cover several MDCT coefficients and more closely match the critical bands of the human ear. This is one of the ways in which MP3 is an improvement over MPEG-1 Layers 1 and 2.

All bands are quantized by dividing by the same value. However, the values in the scale factor bands can be scaled up or down based on their SMR. Bands that have a lower SMR are multiplied by larger scaling factors because the quantization error for these bands has less impact, falling below the masking threshold.

Consider this example. Say that an uncompressed band value is 20,000 and values from all bands are quantized by dividing by 128 and rounding down. Thus the quantized value would be 156. When the value is restored by multiplying by 128, it is 19,968, for an error of $\frac{32}{20000} = 0.0016$. Now supposed the psychoacoustical analyzer reveals that this band requires less precision because of a strong masking tone. Thus, it determines that the band should be scaled by a factor of 0.1. Now we have $floor\left(\frac{20000*0.1}{128}\right) = 15$. Restoring the original value we get $15 * 128 = 19200$, for an error of $\frac{800}{20000} = 0.04$.

An appropriate psychoacoustical analysis provides scaling factors that increase the quantization error where it doesn't matter, in the presence of masking tones. Scale factor bands effectively allow less precision (i.e., fewer bits) to store values if the resulting quantization error falls below the audible level. This is one way to reduce the amount of data in the compressed signal.

6. Encode side information.

Side information is the information needed to decode the rest of the data, including where the main data begins, whether granule pairs can share scale factors, where scale factors and Huffman encodings begin, the Huffman table to use, the quantization step, and so forth.

7. Use Huffman encoding on the resulting 576 quantized MDCT coefficients.

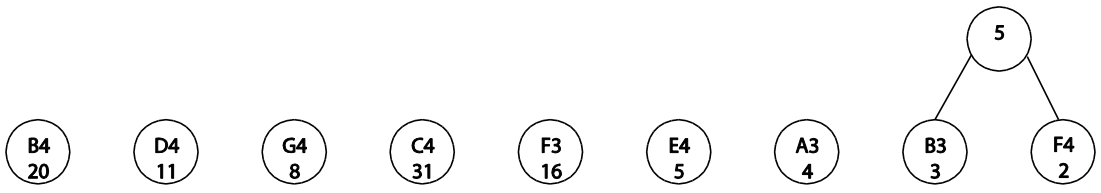
The result of the MDCT is 576 coefficients representing the amplitudes of 576 frequency subbands. Huffman encoding is applied at this point to further compress the signal.

Huffman encoding is a compression method that assigns shorter codes to symbols that appear more often in the signal or file being encoded and longer codes for symbols that appear infrequently. Here's a sketch of how it works. Imagine that you are encoding 88 notes from a piano keyboard. You can use 7 bits for each note. (Six bits would be too few, since with six bits you can represent only $2^6 = 64$ different notes.) In a particular music file, you have 100 instances of notes. The number of instances of each is recorded in the table below. Not all 88 notes are used, but this is not important to the example. (This is just a contrived example, so don't try to make sense of it musically.)

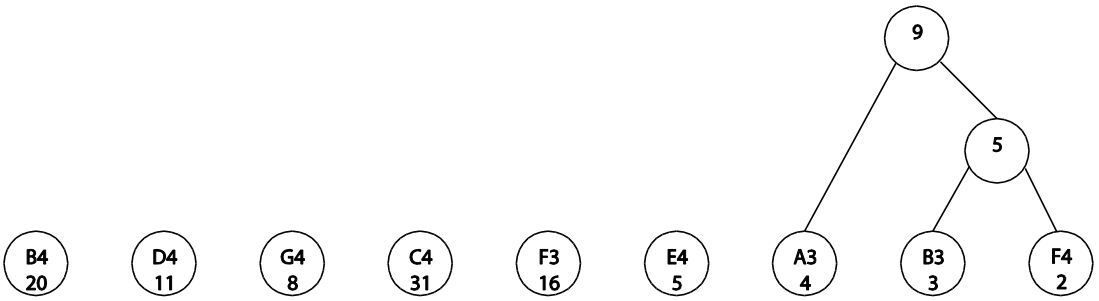
Note	Instances of Note
C4	31
B4	20
F3	16
D4	11
G4	8
E4	5
A3	4
B3	3
F4	2

There are 100 notes to be encoded here. If each requires 7 bits, that's 700 bits for the encoded file. It's possible to reduce the size of this encoded file if we use fewer than 7 bits for the notes that appear most often. We can choose a different encoding by building what's called a Huffman tree. We begin by creating a node (just a circle in a graph) for each of the notes in the file and the number of instances of that note, like this:

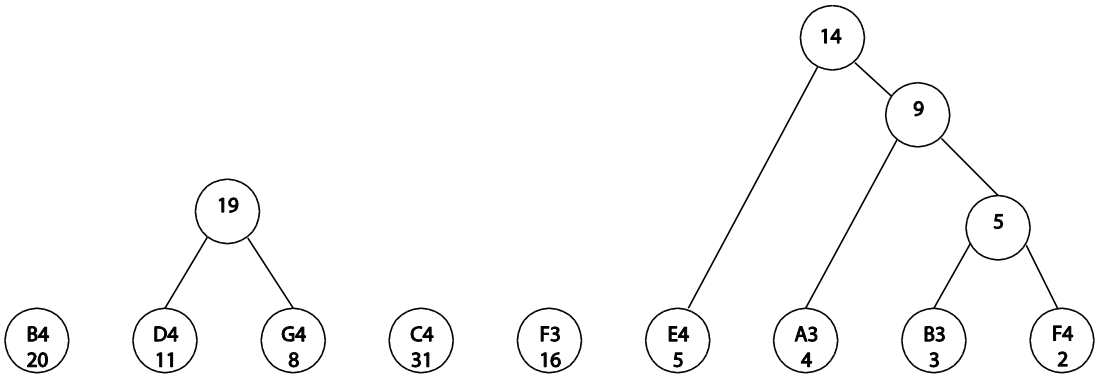
(The initial placement of the nodes isn't important, although a good placement can make the tree look neater after you've finished joining nodes.) Now the idea is to combine the two nodes that have the smallest value, making a node above, marking it with the sum of the chosen nodes' values, and joining the new node with lines to the nodes below. (The lines connecting the nodes are called **arcs**.) B3/3 and F4/2 have the smallest values, so we join them, like this:



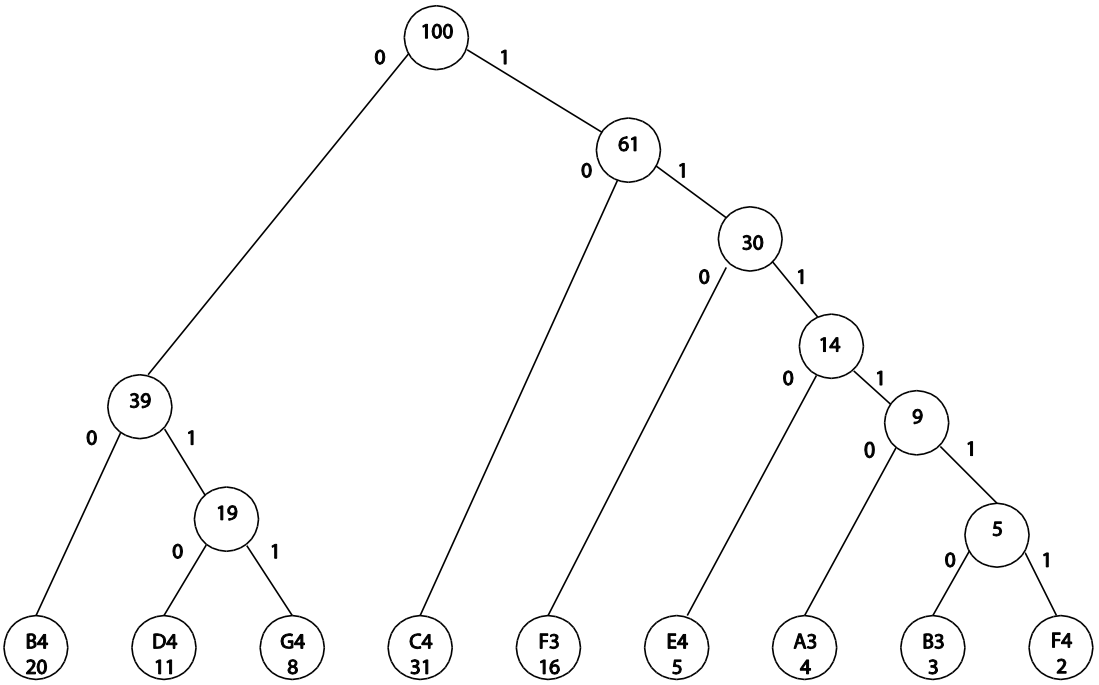
At the next step, A3/4 and the 5 node just created can be combined. (We could also combine A3/4 with E4/5. The choice between the two is arbitrary.) The sum of the instances is 9.



In the next two steps, we combine E4/5 with 9 and then D4 with G4, as shown.



Continuing in this manner, the final tree becomes this:



Notice that the left arc from each node has been labeled with a 0 and the right with a 1. The nodes at the bottom represent the notes to be encoded. You can get the code for a note by reading the 0s and 1s on the arcs that take you from the top node to one of the notes at the bottom. For example, to get to B4, you follow the arcs labeled 0 0. Thus, the new encoding for B4 is 0 0, which requires only two bits. To get to B3, you follow the arcs labeled 1 1 1 1 1 0. The new encoding for B3 is 1 1 1 1 1 0, which requires six bits. All the new encodings are given in the table below:

Note	Instances of Note	Code in Binary	Number of Bits
C4	31	10	2
B4	20	00	2
F3	16	110	3
D4	11	010	3
G4	8	011	3
E4	5	1110	4
A3	4	11110	5
B3	3	111110	6
F4	2	111111	6

Note that not all notes are encoded in the same number of bits. This is not a problem to the decoder because no code is a prefix of any other. (This way, the decoder can figure out where a code ends without knowing its length.) With the encoding that we just derived, which uses fewer bits for notes that occur frequently, we need only 31*2 + 20*2 + 16*3 + 11*3 + 8*3 + 5*4 + 4*5 + 3*6 + 2*6 = 277 bits as opposed to the 700 needed previously.

This illustrates the basic concept of Huffman encoding. However, it is realized in a different manner in the context of MP3 compression. Rather than generate a Huffman table based on the data in a frame, MP3 uses a number of predefined Huffman tables defined in the MPEG standard. A variable in the header of each frame indicates which of these tables is to be used.

Steps 5, 6, and 7 can be done iteratively. After an iteration, the compressor checks that the noise level is acceptable and that the proper bit rate has been maintained. If there are more bits that could be used, the quantizer can be reduced. If the bit limit has been exceeded, quantization must be done more coarsely. The level of distortion in each band is also analyzed. If the distortion is not below the masking threshold, then the scale factor for the band can be adjusted.

8. Put the encoded data into a properly formatted frame in the bit stream.

The header of each frame is as shown in Table 5.4. The data part of each frame consists of the scaled, quantized MDCT values.

AAC compression, the successor to MP3, uses similar encoding techniques but improves on MP3 by offering more sampling rates (8 to 96 kHz), more channels (up to 48), and arbitrary bit rates. Filtering is done solely with the MDCT, with improved frequency resolution for signals without transients and improved time resolution for signals with transients. Frequencies over 16 kHz are better preserved. The overall result is that many listeners find AAC files to have better sound quality than MP3 for files compressed at the same bit rate.

5.3.8.4 Lossless Audio Compression

Lossless audio codecs include FLAC, Apple Lossless, MPEG-4 ALS, and Windows Media Audio Lossless, among others. All of these codecs compress audio in a way that retains all the original information. The audio, upon decompression, is identical to the original signal (disregarding insignificant computation precision errors). Lossless compression generally reduces an audio signal to about half its original size. Let's look at FLAC as an example of how it's done.

Like a number of other lossless codecs, FLAC is based on **linear predictive coding**. The main idea is as follows:

- Consider the audio signal in the time domain.
- Come up with a function that reasonably predicts what an audio sample is likely to be based on the values of previous samples.
- Using that formula on *n* successive samples, calculate what *n+1* "should" be. This value is called the approximation.
- Subtract the approximation from the actual value. The result is called the **residual** (or **residue** or **error**, depending on your source).
- Save the residual rather than the actual value in the compressed signal. Since the residual is likely to be less than the actual value, this saves space.
- The decoder knows the formula that was used for approximating a sample based on previous samples, so with only the residual, it is able to recover the original sample value.

The formula for approximating a sample could be as simple as the following:

$$\sum_{i=1}^{p} \hat{x}(n) = a_i x(n-i)$$

where $\hat{x}(n)$ is the approximation for the n[th] sample and a_i is the predictor coefficient. The residual $e(n)$ is then

$$e(n) = x(n) - \hat{x}(n)$$

The compression ratio is of course affected by the way in which the predictor coefficient is chosen.

Another source of compression in FLAC is **inter-channel decorrelation**. For stereo input, the left and right channels can be converted to mid and side channels with the following equations:

$$mid = (left + right)/2$$
$$side = left - right$$

Values for *mid* and *side* are generally smaller than *left* and *right*, and *left* and *right* can be retrieved from mid and side as follows:

$$right = mid - \frac{side}{2}$$
$$right + side = left$$

A third source of compression is a version of Huffman encoding called **Rice encoding**. Rice codes are special Huffman codes that can be used in this context. A parameter is dynamically set and changed based on the signal's distribution, and this parameter is used to generate the Rice codes. Then the table does not need to be stored with the compressed signal.

FLAC is unpatented and open source. The FLAC website is a good source for details and documentation about the codec's implementation.

5.4 References

Advanced Linux Sound Architecture (ALSA) project homepage. http://www.alsa-project.org/main/index.php/Main_Page.Accessed 08/06/2012.

FLAC. Free Lossless Audio Codec. http://flac.sourceforge.net/. Accessed 09/09/2010.

G.711. "Pulse Code Modulation (PCM) of Voice Frequencies." http://www.itu.int/rec/T-REC-G.711/e. Accessed 09/09/2010.

Hacker, Scot. MP3: *The Definitive Guide*. Sebastopol, CA: O'Reilly, 2000.

ISO/IEC Int'l Standard IS 11172-3:1993. "Information technology -- Coding of Moving Pictures and Associated Audio for Digital Storage Media at up to about 1,5 Mbit/s -- Part 3: Audio." http://www.iso.org/iso/catalogue_detail.htm?csnumber=22412. Accessed 09/08/2010.

Li, Ze-Nian and Mark S. Drew. *Fundamentals of Multimedia*. Upper Saddle River, NJ: Pearson Prentice Hall, 2004.

libsndfile. http://www.mega-nerd.com/libsndfile/. Accessed 08/06/2012.

Rossing, Thomas, F. Richard Moore, and Paul A. Wheeler. *The Science of Sound*. 3rd ed. San Francisco, CA: Addison-Wesley Developers Press, 2002.

4Front Technologies. http://www.opensound.com/. Accessed 08/06/2012.