Physics 1240: Sound and Music Tues 1/24/05

Today: Waveforms and digital sampling

Next Time: Sources of musical sound

Outline

- Audible Range
- Basics of digital sampling
- Sampling rate
- Dynamic range
- Nyquist frequency
- Analysis of waveforms (examples)

Last time:

Frequency (physical) determines pitch (sensation).

(Middle C on the piano keyboard is 261.6 Hz. A half-step is simply a multiple of the twelfth root of 2. We will explain why notes are what they are later in the course. A above middle C is 440 Hz. A half-step is simply a multiple of the twelfth root of 2. We will explain why notes are what they are later in the course.)

Periodicity, period, frequency

Periodic means to repeat at regular intervals.

Only a periodic source produces the sensation of a distinct pitch.

What do we mean by pitch? Pitch is a musical sensation of how high or low a sound is.

frequency = 1 / period

A Hertz (Hz) is a cycle per second.

Audible range is 20 Hz - 20,000 Hz

Vibrations versus waves

Audible range

The frequency range for hearing varies greatly between individuals. The complete audible range is 20-20,000 Hz, but a healthy young person usually can only hear up 17-18 kHz. Our ability to hear the high frequencies continually drops with age. By 55 men can't hear above 5 kHz and women above 12 kHz. Women tend to have better hearing at high frequencies, especially after 25 years of age or so. The ear's sensitivity varies significantly with frequency. We do not hear low frequencies as well as high frequencies, up to around 10 kHz where again our sensitivity drops off. The human ear is most sensitive to frequencies in the range of 2,000-5,000 Hz. Frequencies below 30 Hz are hard to distinguish. Lower frequencies produce more of sense of "feeling" then a sense of sound. We'll say more about these details later in the course.

Digital Sampling

The best way to understand waveforms is to really use them and see them using digital sampling. Let's go through the process of recording a sample starting from the microphone and understand each step.

1) Microphone - There are a variety of microphones, dynamic, condenser, electret condenser. Let's take a look at a dynamic mic. These mics are nice because they don't need an external power feed (either battery or phantom power), they are solid (pretty indestructible) and usually pretty inexpensive. Sound waves cause the pressure to increase/decrease at the mic diaphragm in the audible range (20-20,000 Hz). These increases/decreases in the background pressure push/pull on the diaphragm causing it to move. The diaphragm is connected to a coil of wire, which moves in an external magnetic field supplied by a permanent magnet.

There is a law of physics called "Faraday's Law" that says that if the magnetic field passing through a loop of wire changes, it induces a voltage drop around that loop. In the case of the dynamic microphone, as the coil moves, causing the magnetic field passing through the coil to change. This, in turn, induces a voltage. An electric generator works by using the same principle. The microphone converts a pressure fluctuation into a time varying voltage.

2) The sound card - Now, this voltage from the mic, then goes into the sound card. The sound card has a "preamp" which boosts the time varying voltage (or signal). Next, the signal is converted from a smooth analog signal to a discrete digital signal (represented as integer numbers). You can see visually this sampling process by using Sound Forge to resample at a very low resolution. The soundcard takes discrete samples in time and takes the voltage at each time and converts it to an integer number. The electrical component that converts the analog voltage to a digital voltage (or integer) is called an *analog-to-digital converter*. The whole process is called *digital sampling*.

3) The digital sound editing software then takes this digital signal, stores it displays it and plays it back.

If you are going to do any quality digital recording you are going to need an USB (or firewire) audio interface. These are boxes that take an analog stereo signal, say two mics in, and convert them to a digital signal via the USB. (Especially if you have a laptop.). They are around $200 and available at a music store. They should support 24 bit sound (more on digital sampling next time). Alternately, if you have a mixer you can connect it to audio in on you sound card. My laptop does not even have an audio in. You can buy a cheap USB sound card (there is a SoundBlaster USB interface that is around $40). Though all sound cards take a stereo in 1/8" phono plug. They typically do not support stereo in on the mic jack (Macs actually do).

Summarizing digital sampling, a voltage from the mic, then goes into the sound card. The sound card has a "preamp" which boosts the time varying voltage (or signal). Next, the signal is converted from a smooth analog signal to a discrete digital signal (represented as integer numbers). You can see visually this sampling process by using Sound Forge to resample at a very low resolution. The soundcard takes discrete samples in time and takes the voltage at each time and converts it to an integer number. The electrical component that converts the analog voltage to a digital voltage (or integer) is called an *analog-to-digital converter*. The whole process is called *digital sampling*.

Resample with Soundprobe guitar4.wav at 6000 Hz

Computers use binary integer numbers to represent data. So, we need to know a little (very little) about the conventions used when digital samples are stored on the computer. The quality of a sample is determined by two quantities bit depth and sampling rate. As an example, let's take CD quality recording. There are 44,100 samples taken each second. Or, one every 0.0000227 seconds. Two signals are stored, a left and a right channel. The signal is represented as a "16 bit" binary number. This means in base-ten (the number system we are familiar with working in) the signal runs from:

0 to 2^{16} or from 0 to 65,536 or from -32,768 to +32,768

This is a pretty good quality sample, and it will sound good. We can calculate how much data is involved in storing a CD quality sample for say one second.

44,100 samples x 2 channels x 16 bits = 1,411,200 bits/sec

There are 8 bits per "byte", so

1,411,200 bits / 8 = 176,400 bytes/sec

CDs hold about 74 minutes or 176,400 bytes/sec x 74 minutes x 60 sec/min = 783 Mbytes

Typically, for music, we want a good quality sample (CD resolution or better). You don't always want a CD quality sample rate and resolution because it uses up a lot of memory. Using a digital sound editor, you have the option of 8, 16 or 32-bit depth and 8,000-192,000 Hz sampling rates. Typically, sound cards have CD quality sound capability (or claim to in their specs). My experience is that CD quality is pretty good, and it is hard to hear the difference between 16 bit, 44,100 Hz and a higher quality sample (DAT has a 48 kHz sampling rate). Typically, audio professionals prefer 24 bit samples at sampling rates faster than 44,100 Hz (96 or 192 kHz) because quality always degrades when