PHYSICS OF MUSIC

REFERENCES:

1.) Charles Taylor, *Exploring Music* (Music Library ML3805 T225 1992)

2.) Juan Roederer, *Physics and Psychophysics of Music* (Music Library ML3805 R74 1995)

3.) Physics of Sound, writeup in this lab manual

4.) J. Lattard, *Gammes et Temperaments musicaux* (Music Library ML3809 L24 1988)

5.) Cool Edit manual (available in each sound room)

OVERVIEW OF AVAILABLE EQUIPMENT

1.) PC interfaced to cassette tape player, CD player, and microphone: allows you to analyse live or recorded music. So if you don't feel like bringing your double bass into the lab, just make a tape of what you'd like to analyse!

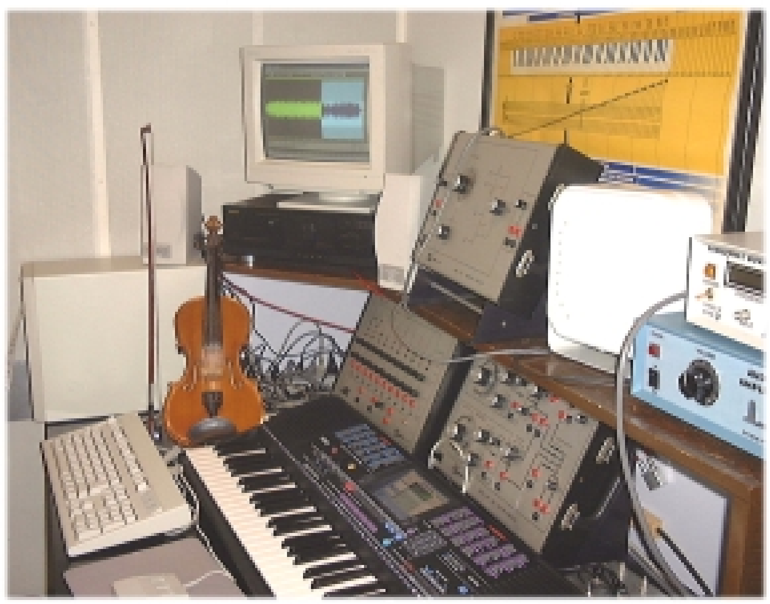

Figure 1

2.) Dual channel Analog Signal Generator, Analog Fourier Synthesizer: apparatus for combining up to 9 analog sine, triangle, or square waves.

3.) Yamaha Synthesizer/Keyboard with MIDI interface: generates a wide variety of synthesized and sampled sounds.

4.) *Cool Edit* software for PC: allows (i) detailed frequency and amplitude analysis of sounds (ii) addition of special effects (eg. echo, reverberation, envelopes, etc.), and (iii) digital generation of sounds (eg. sine, triangle, and square waves, noise, etc.) (iv) cutting and pasting of audio clips

PREREQUISITES:

This experiment relies heavily on digital sound processing software. You should be familiar with the Windows environment before attempting the experiment. If you have no prior experience with digital signal processing please make sure you read the short selection in the Cool Edit manual entitled “A Short Course in Digital Sound Processing”.

The experiment also requires a bit of background knowledge in music. Ideally you should play a musical instrument and be familiar with basic musical theory terms (eg. perfect fifth, key signature, semitone, middle C).

INTRODUCTION:

There is a tremendous variety of experiments that can be done with the equipment in the Physics of Music laboratory. References 1 and 2 above contain discussions of many possible experiments. Please look through these books for ideas and discuss what you would like to do with your demonstrator or with a lab coordinator. Alternatively, we outline a few basic experiments below.

Regardless of what you choose to do in this lab, it is almost certain that you will, at some point, need to measure the frequency of sounds very precisely. Therefore, before doing anything please complete the following short exercise:

Preliminary exercise: Frequency measurements

Measure the frequency of a tone using the following two methods. Do your two results agree? Which method is better? Can you suggest a reason why?

<u>Method 1</u>

Record a sound. Then select and zoom in on a selected region that is small enough for you to see individual oscillations. Count the number, N, of complete oscillations in this region, and measure the duration, T, of this region. The frequency, f, is just N/T. Calculate the uncertainty in f based on your estimate of the error in N and T.

<u>Method 2</u>

This method uses a numerical algorithm called a Fast Fourier Transform to determine the frequency components in a sound. (See the Physics of Sound lab writeup for some background information on Fourier analysis.)

Select a region of recorded sound, then type alt+z (or select *frequency analysis* from the *options* menu bar item). A graph of the frequency distribution will appear, along with a listing of the loudest frequency component in the left and right channel. You may take the reading error in this frequency to be one half of the last place quoted. Measure the frequency in several nearby regions to estimate the standard deviation.

NOTE: You may find that you can reduce reading errors slightly by reducing the sampling rate of the digital recording software. (eg use a sampling rate of 8000 instead of 44000 Hz.) However be sure to remember that the maximum frequency a digital system can record is one half the sampling rate.

<u>EXPERIMENT SUGGESTIONS</u>

Experiment 1: OVERTONES

The notes generated by most acoustic instruments are generally not simple sine waves. Instead the notes are made up of several different sine wave components, or overtones. Provided the instrument produces a relatively steady tone, there is usually a simple relationship between the frequencies of these overtones: the frequency of all the overtones is an integer multiple of the lowest frequency component, the fundamental. Thus if one plays a note of fundamental frequency 440 Hz on a guitar, components of frequency 880Hz, 1320 Hz, and so on, will also be present. Can you suggest a physical reason why this might be so?

Determine the amplitude and frequency of overtones present in the sounds produced by a number of musical instruments. (Use the alt+z frequency analysis function to produce a graph. Then use the mouse to move the cursor around and read off peak locations.) Plot your data to determine if the overtone frequencies are all an integer multiple of a common fundamental frequency. On the same plot indicate the amplitude of the various overtones.

Curiously enough, the human ear-brain system is capable of determining what the fundamental frequency is even if it is not present in the overtone series! Try to verify this for yourself: use either the analog Fourier synthesizer, or the *tones* function in the Cool Edit *generate* menu item to generate a tone made up of the components f_o, $2f_o$, $3f_o$ ....etc.. Then generate the tone again, but this time with the amplitude of the f_o component set to zero. What pitch does your ear perceive?

Some notes on some acoustic instruments (eg. violin, oboe) make use of this phenomenon. Look at the overtone structure for a number of different notes in the instruments available to you and try to find cases where the amplitude of the fundamental is small. Changes in embouchure (for woodwinds and brasses) or bowing speed and pressure (for the strings) can change overtone structure substantially. Try describing some of these changes qualitatively. (eg. What conditions lead to particularly small or large fundamentals?)

Experiment 2: BEATS

If two sine waves of equal amplitude, A, but of different frequencies, f_1 and f_2, are added together:

$$\phi_{tot} = A\sin(2\pi f_1 t) + A\sin(2\pi f_2 t)$$

the result is, using an elementary trigonometric identity,

$$\phi_{tot} = 2A\cos[2\pi t(f_1-f_2)/2] \times \sin[2\pi t(f_1+f_2)/2] \qquad \text{... (1)}$$

A plot of this function is shown in figure 1. The high frequency oscillations correspond to the sine term, and have a frequency of $(f_1+f_2)/2$, which is just the average frequency of f_1 and f_2. This is the pitch your ear hears. The amplitude of the waveform is determined by the much more slowly varying cos term which has a frequency of $(f_1-f_2)/2$. The periodic changes in amplitude at this latter frequency are called beats.

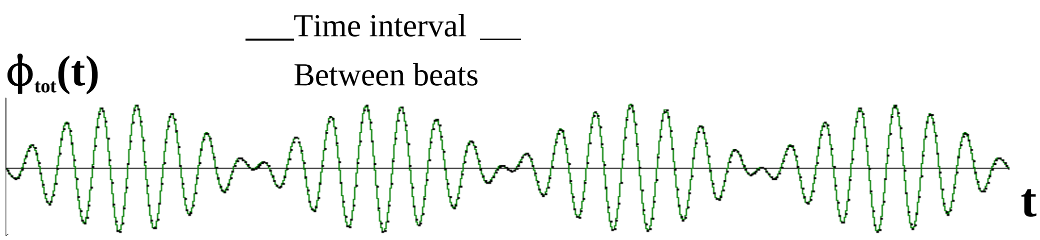

Figure 1. The sum, $\phi_{tot}(t)$, of two sin waves (equation 1) of similar frequency.

Combine two sine waves and listen to the result by using the *tones* function in the Cool Edit *generate* menu item. To see and hear beats you will need two frequencies relatively close to each other (say 1.00 and 1.01 times the fundamental frequency). Check to see if the main and beat frequencies are consistent with the above equations.

If you have access to a stringed instrument play the same note on different strings simultaneously. Slightly detune one of the notes until you hear beats. Try to record the result and measure the beat frequency.

When two tones from a musical instrument are sounded it is not just the fundamentals that will beat, but all of the overtones as well. Based on table 1 what should the beat frequency be between C_4 and G_5? (C_4 is middle C, G_4 is the G above middle C, G_5 is one octave above G_4 and so on.) Try to measure the beat frequency for this or other intervals on the Yamaha keyboard and compare with what you expect.

C_4	D_4	E_4	F_4	G_4	A_4	B_4	C_5
261.6	293.6	329.6	349.2	392	440	493.9	523.2

Table 1. Frequencies (in Hz) of the notes in the C major scale. (Equal tempered tuning)

Experiment 3: MUSICAL SCALES

A musical scale is a set of tones that blend together in an "aesthetically pleasing" way. What this means is that there are at least as many different scales in existence as there are different cultures! Two types of scales that have seen a good deal of usage in occidental music are the Just Diatonic and the Equal Tempered scales. These two scales are based largely on the premise that "aesthetically pleasing" means that beating between notes (and their overtones) of the scale should be as small as possible.

The Just Diatonic Scale

The Just Diatonic scale, which came into being in the 17th century, is based on two fundamental intervals: the octave (frequency ratio 2:1) and the perfect fifth (frequency ratio 3:2). ("perfect" simply means that the frequency ratio corresponding to the interval is a simple fraction.)To get the notes in the Just Diatonic scale we take a base note, the notes which are a perfect fifth above and below this base note, and the first five overtones of these three notes. These notes cover a very wide range of frequencies. To bring all of the notes into the octave above our base note, we successively divide or multiply their frequencies by 2. It turns out that when this is done (see reference 3), and the resulting notes are arranged in order of increasing pitch we arrive at the frequency ratios shown in table 2. Table 2 also shows the resulting note frequencies when C_4 is taken as the base note. Because all the notes in this scale are derived from the overtones of three harmonically related notes, beating between notes of the scale is minimal. The scale has a serious drawback however: if we use the same process as described above to create a scale with a base note other than C_4 (D_4 for example) we get a series of notes whose frequencies are different from the notes in the C_4 scale. Thus if one uses the Just system every key signature requires a differently tuned instrument. A keyboard using the Just system would require hundreds of keys!

	C_4	D_4	E_4	F_4	G_4	A_4	B_4	C_5
frequency ratio relative to base note	1	9/8	5/4	4/3	3/2	5/3	15/8	2
frequency (Hz) of Just Diatonic scale notes (based on A_4= 440Hz)	264	297	330	352	396	440	495	528

Table 2. Frequencies and frequency ratios (relative to the base note) for notes in the Just Diatonic scale.

Try playing this scale by using the *generate tones* menu item of Cool Edit. Does it sound different from the diatonic scale you are use to? If so, try to describe the differences qualitatively.

The Equal Tempered Scale

The Equal Tempered scale is a 12 note scale from which Diatonic scales may be constructed in any key. Thus a keyboard based on the Equal Tempered scale requires far fewer keys than a Just system keyboard. The main disadvantage of the equal tempered tuning is that the only perfect interval in this scale is the octave. All other intervals are slightly mis-tuned. This led to a great deal of discontent when the scale was first introduced. For example, in *The Philosophy of Music* (1879), William Pole wrote:

> The modern practice of tuning all organs to equal temperament has been a fearful detriment to their quality of tone. Under the old tuning an organ made harmonius and attractive music, which it was a pleasure to listen to. . . . Now, the harsh thirds, applied to the whole instrument indiscriminately, give it a cacophonous and repulsive effect.

Nevertheless virtually all keyboards today (including the Yamaha synthesizer in this lab) use equal temperament tuning.

The Just diatonic scale discussed above may be thought of as the following series of intervals: T T S T T T S where T stands for "tone" and S stands for "semitone". The frequency ratio corresponding to a tone is roughly twice that corresponding to a semitone. Thus an octave spans a total of 12 semitones. The equal tempered system assigns a fixed frequency ratio, R_s, to each of these semitones. Thus the frequency of a note one semitone above a note of frequency f_0 is $f_1 = R_s f_0$. The frequency of a note two semitones above f_0 is $f_2 = (R_s)^2 f_0$ and so on. In general, a note which is n semitones above f_0 has a frequency given by:

$$f_n = (R_s)^n f_0.$$

Since an octave corresponds to a frequency ratio of 2:1, $(R_s)^{12} = 2$, or

$$R_s = \sqrt[12]{2}$$

Thus a plot of $\ln(f_n)$ vs n should yield a straight line. What should the line's slope be? Try measuring and plotting the frequency of a number of synthesizer notes to test your prediction. Now sing or play a scale on an acoustic instrument (not a keyboard) and make a similar plot. Which fit yields a better χ^2. Why? Alternatively, you may find it interesting to compare the pitches used by professional musicians with the equal tempered scale. Keyboard pitches will naturally be close to equal tempered. However vocalists, string players and some woodwind and brass players often use perfect intervals. You can use Cool Edit to record and frequency analyse material recorded on cassette tape and/or CD.

Experiment 4: TRANSIENT START-UP and DECAY of TONES

A large body of research (see reference 2 for an overview) suggests that one of the most important characteristics of musical instrument tones is the envelope that defines their start-up and decay. It is found that the first fraction of a second of a note is what allows the human ear to distinguish between say a piccolo and a violin, or a tuba and a piano. There are plenty of experiments that can be done involving transients. Try to think up a few yourself, or try the suggestions below:

(i) Use Cool Edit to qualitatively explore the difference in startup transients between different sounds. How long does it take to go from the start-up of a note to a steady tone? Does the note decay exponentially? If so what is the time constant? Measure the envelope shapes and/or time constants of a variety of percussive sounds.

(ii) Try "gluing" the startup of a clarinet note to the steady state tone generated by, say, an electric guitar. (Use *cut* and *paste* in Cool Edit's *edit* menu item.) What instrument does the resulting note most closely resemble? Do this for a variety of combinations and come to some conclusion about what is more important: the start-up transient, or the steady state tone.

(iii) How does the overtone structure of a note in the startup region compare with the overtone structure in the steady tone region? (This will depend radically on what type of instrument you're looking at!)

(mf - 97)