

PHYSICS OF SOUND

REFERENCES

Acoustics:

1. M.M. Sternheim and J.W. Kane. *General Physics*. (Second edition), John Wiley & Sons, Toronto, 1991, pp. 564-568.

Instruments:

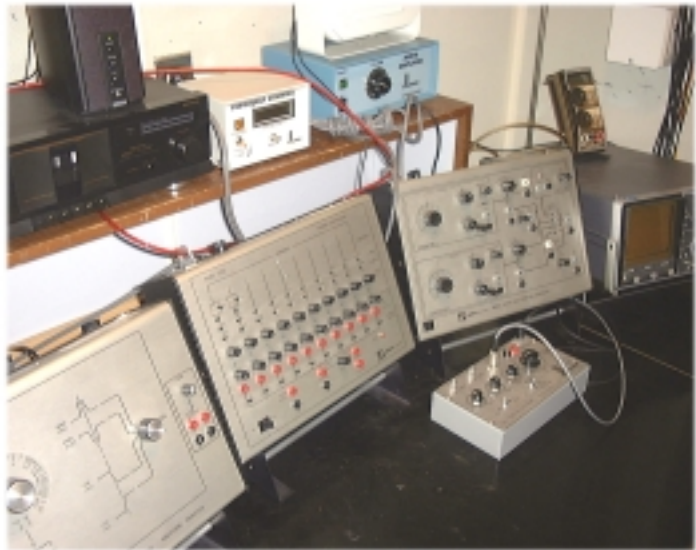
1. Sections "The Multimeter" and "The Oscilloscope" of this Lab Manual's chapter "Commonly Used Instrument".
2. Instrument Specifications (available on the computer).
3. User's manual for the various instruments (available at the **Resource Centre**).

Circuit-Wiring:

Chapter "Circuit-Wiring Technique" of this Lab Manual.

INTRODUCTION

This is a six part package on the Physics of Sound, Music, and Sound Reproduction. You may attempt whichever parts of the experiment that you wish, including parts that are not described here but which interest you. Successful completion of Experiments 1 through 3 constitute two weights: completion of all six experiments would constitute four weights. The six sections of the experiment for which guide sheets are prepared are:



●*Experiment 1: Frequency, Pitch & Decibels:* An investigation of some basic concepts of the Physics of Sound, and an introduction to the apparatus. We strongly recommend that at least the basic concepts discussed in this section are familiar to you before attempting further experimentation.

●*Experiment 2: Addition of Waves I - Two Waves:* Topics include Beats, Amplitude Modulation, and Frequency Modulation. There is no prerequisite, but we expect familiarity with the concepts discussed in Experiment 1.

●*Experiment 3: Addition of Waves II - Fourier Analysis:* Fourier's Theorem, Overtones, Harmonic Analysis, Phase Shift and Non-Linear Circuits are discussed. We expect you to have spent a few hours doing Experiment 1 and/or Experiment 2.

●*Experiment 4: Harmonic Analysis and Synthesis:* A direct extension of Experiment 3 which is a prerequisite. Here musical and other sources are analyzed for harmonic structure, and then synthesized.

●*Experiment 5: Loudspeaker Principles:* Loudspeaker design is more "art" than "science". In this experiment some of the parameters of this art are investigated. Topics include Frequency Response, Dispersion, Impedance, and Tone-Burst Response. Prerequisite is a few hours doing Experiment 1 and/or Experiment 2. You may bring in your own loudspeakers to study if you wish, at your own risk!

●*Experiment 6: Sound Levels:* Measuring sound levels found in the environment. We expect you to be familiar with the concepts discussed in Experiment 1.

Each of these experiments is open-ended, and can lead you into a variety of topics including Wave Theory, Electronics, Psycho-acoustics, and a great deal else. Thus, you may find this package more challenging than the usual First Year Lab experiment. Also, the equipment in the package is quite versatile, and a great deal more than these 6 experiments may be investigated in consultation with your demonstrator.

Although many parts of this experiment require careful measurement and analysis of data, there are some parts in which the primary concern is with what things sound like to you.

Much of this package is electronic in the sense that electrical signals are being dealt with until the very last stage, the loudspeaker, where the electrical signal is converted into a sound wave. Take your time with the wiring, so that you may avoid "not being able to see the principles for the wires".

These experiments are done in the semi-anechoic chambers, Rooms 258, 259 and 260. Keys for the locks are available at the **Resource Centre**. Not only are these rooms sound proofed, but they are also designed to absorb most of the sound energy incident on the walls and ceiling. This is the property that makes these chambers sound "dead". You may wish to ponder the fact that in the average room, 60% of the sound energy you hear from your record player has been reflected off the wall, ceiling and floors of the room.

Do not leave the padlock in the door of the room when you are inside; take the lock inside with you.

EXPERIMENT 1: - FREQUENCY, PITCH & DECIBELS

INTRODUCTION

This experiment concentrates on some of the basic concepts used in the study of the physics of sound. Thus, it is the "simplest" of the experiments in the package. However, it does talk about a fairly large number of concepts and definitions which may be unfamiliar to you, and a careful experimental study of all the sections of this guide sheet may take a large amount of time. We strongly recommend that you be familiar with at least the basic concepts discussed here before attempting further experiments.

APPARATUS:	Dual Function Generator (DFG)	Controller
	Amplifier	Speaker
	Oscilloscope	Multimeter

SETTING UP:

Set the controls of the Dual Function Generator (DFG) as follows:

- 1) Both MODULATION switches OFF
- 2) All 4 SUMMING AMPLIFIER switches to OUT
- 3) All 3 GAIN controls turned down
- 4) Both AMPLITUDE controls turned down
- 5) Both NOR/INV switches to NOR
- 6) Both FREQUENCY controls to $\approx 300\text{Hz}$ (Dials full clockwise, range on 300)
- 7) Sine wave shape for both generators

Set the controls of the controller as follows:

- 1) Switch position to no. 1
- 2) TAPE switch to "Dual Function Generator"
- 3) All 3 GAIN controls fully on
- 4) MASTER VOLUME control fully down

Wire the OUTPUT from GENERATOR 1 of the DFG to the 10K INPUT 1 of the controller. Similarly wire the OUTPUT from GENERATOR 2 of the DFG to 10K INPUT 2 of the controller. Wire the METER jacks on the controller to the oscilloscope. Turn on the oscilloscope. Make sure that the TO AMPLIFIER jack on the controller is wired to the input of the amplifier. Now plug in the amplifier power cord and turn on both the DFG and the amplifier. The last section of the guide details how to turn this system off.

FREQUENCY AND PITCH:



Turn up the AMPLITUDE from generator 1, until you get about 0.5 volts p-p on the oscilloscope. ("p-p" means peak-to-peak, and means the voltage from the most negative part of the pulse to the most positive part. See the figure.



You may measure the period T of the signal with the oscilloscope, and confirm that the frequency $f=1/T$. You may also *hear* this tone by turning up the MASTER VOLUME control. Listen to what happens when the frequency of the tone is changed. Also notice the changing period on the oscilloscope. The maximum range of human hearing is approximately 20Hz to 20kHz, although our loudspeaker does not quite make these extremes.

(This maximum range is approximately true for young children.) Check *your* own hearing frequency range. Depending on your age and the state of your hearing you will find your frequency maximum is below the 20kHz. For comparison the lowest note on an organ is 32Hz. The 20th century standard "A" above middle "C" is 440Hz (In the baroque period, pitch was not standardized and instruments were tuned to "A"s ranging from about 400Hz to about 490Hz, a spread of about two whole tones.) The highest note on a piano is 4.186kHz. The "whistle" from the horizontal oscillator of television sets is at 15.750kHz. (If you don't hear that annoying T.V. whistle, you know your hearing drops off below that frequency.)

Also notice that for any tone, the frequency of the tone one octave above the original is double the original frequency, and the frequency of the tone one octave below the original is one-half the original frequency. The frequency ratio of two notes at an interval of a "fifth" is 3:2, and of a "major third" is 5:4.

The **timbre** of the tone can be changed by switching to either the triangular wave  or the square wave . You will perceive the change in timbre more pronouncedly for frequencies below 3kHz. This effect is further investigated in Experiment 3.

DECIBELS AND LOUDNESS:

The ratio of sound level intensities, or power levels $\frac{P_1}{P_2}$ to which the ear can respond, from the just perceptible to the threshold of pain, is 10^{12} . Further, the ear responds to differences in intensity in a non-linear approximately logarithmic fashion.

Thus, the physics of music uses a log scale for comparing two sound intensities: if two sound intensities are P_1 and P_2 , the difference in the two intensities, in decibels, is

$$dB = 10 \log_{10} \frac{P_1}{P_2} \quad \text{where } P \text{ is the intensity or power}$$

Common Sound Levels

	P (μ Watts/ m^2)	Level (dB)
Threshold of Hearing	10^{-6}	0
Rustling Leaves	10^{-4}	20
Talking at 3 ft	10^{-2}	40
Noisy office	1	60
Subway car	10^4	100
Loud Rock Band	10^5	110
Threshold of Pain	10^6	120

Notice that with the dB scale which level you choose to be 0 dB is arbitrary since all other levels are compared to the one reference level. However, in psycho-acoustics and many other measurements 0 dB is chosen to be exactly $10^{-6} \mu W/m^2$.

Often one measures voltages instead of intensities or power levels. Since the power varies as the square of the voltage:

$$P = \frac{V^2}{R}$$

The same differences in intensities in terms of the voltage is

$$dB = 10 \log_{10} \frac{V_1^2}{V_2^2} = 20 \log_{10} \frac{V_1}{V_2} \quad \text{where } V \text{ is the voltage}$$

The minimum perceptible change in intensity which the ear can detect varies with pitch and intensity level, and is in the range of 0.2 to 8 dB. You may measure this by setting the frequencies of both generators of the DFG to be equal and adjusting the AMPLITUDE so that

you can just perceive the difference in level when switching back and forth between 1 and 2 on the controller. More accuracy may be achieved by using the multimeter at the METER output to measure the A.C. voltage. The multimeter measures *rms* voltage, not peak-to-peak voltage, so don't expect the meter to duplicate the oscilloscope readings. In fact:

$$V_{rms} = \frac{V_{p-p}}{2\sqrt{2}} = 0.354V_{p-p}$$

TURNING THE SYSTEM OFF:

Follow these instructions in order:

- 1) Turn all GAIN, AMPLITUDE and VOLUME controls to low.
- 2) Turn off all power switches.

EXPERIMENT 2: - ADDITION OF WAVES PART I: - Two Waves

"Wherever we are, what we hear is mostly noise. When we ignore it, it disturbs us. When we listen to it, we find it fascinating."

... John Cage

INTRODUCTION:

In this experiment various ways of combining two waves are explored. There is no prerequisite as such, but a general familiarity with the concepts discussed in Experiment 1, "Frequency, Pitch & Decibels" is assumed.

APPARATUS:	Dual Function Generator (DFG)	Controller
	Amplifier	Loudspeaker
	Oscilloscope	Frequency Counter

SETTING UP:

Set the controls of the Dual Function Generator (DFG) as follows:

- 1) Both MODULATION switches OFF
- 2) All 4 SUMMING AMPLIFIER switches to OUT
- 3) All 3 GAIN controls turned down
- 4) Both AMPLITUDE controls turned down
- 5) Both NOR/INV switches to NOR
- 6) Both FREQUENCY controls to $\approx 300\text{Hz}$ (Dials full clockwise, range on 300)
- 7) Sine wave shape for both generators

Set the controls of the controller as follows:

- 1) Switch position to no. 1
- 2) TAPE switch to "Dual Function Generator"
- 3) All 3 GAIN controls fully on
- 4) MASTER VOLUME control fully down

Wire the 10K OUTPUT from the SUMMING AMPLIFIER of the DFG to the 10K INPUT 1 of the controller. Wire the METER jacks on the controller to the oscilloscope. Turn on the oscilloscope. Make sure that the TO AMPLIFIER jack on the controller is wired to the input of the amplifier. Now turn on both the DFG and the amplifier.

BEATS:

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

$$\Psi_{tot} = A \sin 2\pi f_1 t + A \sin 2\pi f_2 t$$

the result is:

$$\Psi_{tot} = 2A \left[\cos 2\pi \frac{f_1 - f_2}{2} t \right] \times \left[\sin 2\pi \frac{f_1 + f_2}{2} t \right]$$

Examine the right hand sine term. The average frequency is just:

$$f_{av} = \frac{f_1 + f_2}{2}$$

so:

$$\Psi_{tot} = 2A \left[\cos 2\pi \frac{f_1 - f_2}{2} t \right] \times \left[\sin 2\pi f_{av} t \right]$$

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Thus, the total wave has a frequency equal to the average, but the amplitude changes with time

according to $\cos 2\pi \frac{f_1 - f_2}{2} t$. These variations in amplitude are called beats.

Switch the output of GENERATOR 1 to IN the SUMMING AMPLIFIER. Now adjust the AMPLITUDE from the GENERATOR 1 to about 0.5 volts p-p as seen on the oscilloscope. Switch GENERATOR 1 to OUT, and switch GENERATOR 2 to IN and adjust its AMPLITUDE for the same voltage. Now switch IN both generators, and observe the result on the oscilloscope, looking at this summed signal using a variety of horizontal sweep speeds. You may also hear the result by turning up the MASTER Volume control.

Slowly vary the frequency on one generator and observe the results. If the two frequencies f_1 and f_2 are close to each other, you may measure the period of the beats by ear with a stopwatch. If they are further apart the period may be measured with the oscilloscope. In either case you will want to measure the frequencies from the two generators, f_1 and f_2 , with a frequency counter. To measure f_1 , insert the lead from the counter in the TRIG OUTPUT from GENERATOR 1; to measure f_2 insert the lead from the counter into the TRIG OUTPUT from GENERATOR 2. Compare the frequency of the beats with $f_1 - f_2$.

What happened to the number 2 in the term $\frac{f_1 - f_2}{2}$?

When a musician is trying to tune an instrument with reference to another, a common technique is to use the elimination of beats as a reference. The lowest two notes on a piano are A_o (27.5 Hz) and $A_o^\#$ (29.1 Hz). Their beat frequency is thus 1.6 Hz, which is easily detectable. If you have access to a piano, hit both notes together gently. Is the piano in tune?

AMPLITUDE MODULATION:

When you change the amplitude control on the generators, you are "modulating" the amplitude. In the section on beats we saw a way to take a wave of frequency f_{av} and modulate its amplitude with another of frequency, Δf . The DFG allows for this directly. Set GENERATOR 2 to about 300Hz, and GENERATOR 1 to a frequency between 1 and 50Hz. Switch GENERATOR 1 OUT of the SUMMING AMPLIFIER. Now, switch the lower MODULATION switch to AM, and listen to the result as you vary the AMPLITUDE and frequency of GENERATOR 1. To see this on the oscilloscope, you will probably want to use the external trigger on the oscilloscope connected to the TRIG OUTPUT of GENERATOR 1.

A wide range of "musical" effects (noise effects?) occur when both the modulation frequency and the frequency of the signal which is modulated are in the range of audibility. And, of course, the modulation wave shape need not be a simple sine wave.

This is what the AM means in AM radio.

FREQUENCY MODULATION:

You may also use the output of GENERATOR 1 to change, *i.e.*, "modulate", the frequency of GENERATOR 2. Switch the lower MODULATION switch to FM. The frequency of GENERATOR 1 controls the frequency of the changing frequency from GENERATOR 2, the AMPLITUDE of GENERATOR 1 controls the amount of modulation that occurs.

Musical vibrato is usually a frequency modulation of 5 to 7Hz and small amount. A large amount of FM at this frequency makes a police car siren. The internal trigger of the oscilloscope often works best in displaying FM. The other two types of modulation available are:

DSB: double side band. This is simply amplitude modulation with the carrier frequency suppressed.
GATE - The output of generator 1 turns generator 2 on and off. This may be used to generate tone bursts. See Experiment 5.

TURNING THE SYSTEM OFF:

Follow these instructions in order:

- 1) Turn all GAIN, AMPLITUDE and VOLUME controls to low.
- 2) Turn off all power switches.

EXPERIMENT 3: - ADDITION OF WAVES II - Fourier Analysis

Reference: Armstrong & King, Mechanics, Waves & Thermal Physics, pg. 330 ff.

Feynman, Leighton & Sands, The Feynman Lectures on Physics, Vol. 1, Lect. 50.

INTRODUCTION:

In this experiment the subject of Fourier analysis is investigated. We expect you to be familiar with the concepts discussed in Experiment 1: "Frequency, Pitch, and Decibels", and to have spent a few hours doing Experiment 1 and/or Experiment 2 from this set on the Physics of Sound. In addition, those with the necessary mathematics will find the references useful in further understanding the theory behind this experiment.

APPARATUS:	Dual Function Generator (DFG)	Controller
	Amplifier	Speaker
	Oscilloscope	Multimeter
	Fourier Synthesizer	Waveform Analyzer
	Frequency Counter	

SOME THEORY:

Consider a vibrating string of length L . The possible wavelengths for resonant standing waves on this string are:

$$\lambda_{allowed} = \frac{2L}{n} \quad n = 1,2,3,4,\dots$$

Thus, there is an infinite number of possible standing waves. Then $n = 1$ standing wave is called the fundamental; it is the note to which the string is tuned. The frequency of the sound wave from this wave is $f_0 = \frac{v}{\lambda} = \frac{v}{2L}$ where v is the velocity of the wave down the string.

The waves for $n > 1$ are the overtones of the string, and it is the relative amounts of these tones that determines the timbre of the note. The frequencies of these overtones, f , are equal to nf_0 .

The point is that an actual string will vibrate in a complicated manner, but the complicated manner is just the sum of the vibrations of the fundamental and the overtones. And, in fact, the mathematicians have proved that any complicated periodic wave we're likely to run into can be written as a sum of simple sine and cosine terms.

This is called Fourier's Theorem and we state it for reference:

Given a periodic wave $F(t)$ with period $T = \frac{1}{f_0}$ with a finite number of discontinuities, extreme values, maxima and minima, then:

$$f(t) = \sum_{n=0}^{\infty} [A_n \cos 2\pi n f_0 t + B_n \sin 2\pi n f_0 t]$$

where

$$A_n = \frac{2}{T} \int_0^T f(t) \cos 2\pi n f_0 t \cdot dt \quad B_n = \frac{2}{T} \int_0^T f(t) \sin 2\pi n f_0 t \cdot dt$$

SETTING UP:

Set the controls of the Fourier synthesizer as follows:

- 1) All-AMPLITUDE and GAIN knobs turned down
- 2) All SUMMING AMPLIFIER switches OUT
- 3) Power OFF

Set the controls of the controller as follows:

- 1) Switch position to no. 1
- 2) TAPE switch to "Dual Function Generator"
- 3) All 3 GAIN controls fully on
- 4) MASTER VOLUME control fully down

Wire the 10K OUTPUT from the SUMMING AMPLIFIER of the Fourier synthesizer to the 10K INPUT 1 of the controller. Wire the METER jacks on the controller to the oscilloscope; Connect the TRIGGER OUTPUT from the synthesizer to the external trigger input on the oscilloscope. Also wire the multimeter across the METER jacks to measure A.C. voltage. Turn on the oscilloscope and the meter. Finally, turn the main amplifier and the synthesizer ON.

SAWTOOTH WAVE:

To demonstrate the principles of Fourier analysis, consider a sawtooth wave



$$f(t) = 2 \pi f_o t \quad \text{for } -\pi < 2\pi f_o t < \pi$$

Such a wave-shape is available on the Dual Function Generator. The Fourier series of this wave mathematically works out to be:

$$f(t) = 2[\sin 2\pi f_o t - \frac{1}{2}\sin 4\pi f_o t + \frac{1}{3}\sin 6\pi f_o t - \dots] = \sum_{n=1}^{\infty} 2 \frac{(-1)^{n+1}}{n} \sin 2\pi n f_o t$$

To set this wave up on the synthesizer, proceed as follows. First select either of the two identical channel 1's. Switch the summing AMPLIFIER to IN for that channel, and select the sine wave shape. Touch the RESET button, and then adjust the AMPLITUDE for that channel to between 8 and 10, and note the voltage V_o , on the multimeter. Now, find a combination of phase switches: 0° , 90° , 0° , 180° , and VARIABLE PHASE, so that on the oscilloscope the wave is a sine wave, *i.e.*, starts at zero amplitude and is increasing.

(The convention used in the Fourier synthesizer is that the trigger output gives a pulse to synchronize with a cosine wave, so that to have the oscilloscope showing a sine wave, the phase must be set at 90° .) This is the $n = 1$ term in the series, and has a frequency $f_o = 440\text{Hz}$ (international "A") and amplitude = V_o . The following table lists the frequencies for each channel of the Fourier synthesizer.

<u>Channel</u>	<u>Frequency</u> <u>(Hz) $\pm 0.1\%$</u>
1	440
2	880
3	1320
4	1760
5	2200
6	2640
7	3080
8	3520
9	3960

Now, temporarily switch channel 1 OUT without disturbing the controls and switch channel 2 IN. Again, set the RESET. This is $n = 2$ term in the series with frequency = $2f_o$ and should be adjusted so that the AMPLITUDE is $\frac{1}{2}V_o$, and the phase is a negative sine wave, *i.e.*, starts at zero amplitude and is decreasing. Now switch channel 2 OUT, and switch channel 3 IN. Set the RESET. Examination of the Fourier series above shows that the amplitude is $\frac{1}{3}V_o$, and the phase is a positive sine wave. Continue in this manner until all 9 channels have been set up in accordance with the Fourier series. It is a good idea to hit the RESET button before each channel is adjusted to keep transients in the power line from affecting the digital circuitry and changing phase relationships. Now switch channel 1 IN and add the harmonics one by one, and observe that the wave gets closer and closer in shape to the desired sawtooth shape.

Of course, the representation will not be perfect because we have not added an infinite number of terms. In particular, the series will always overshoot at any discontinuity by about 18%, but as terms are added the width of the overshoot decreases; this is called Gibbs phenomenon.

One way of presenting your data is to take a Polaroid photograph of the oscilloscope as various terms are added to the series. A camera is available at the Resource Centre. You may wish to try varying the AMPLITUDE and phase of the harmonics slightly to produce a better representation.

You may also compare the sound of the synthesized wave to the sawtooth wave produced by the Dual Function Generator (DFG). Simply wire the output of one generator of the DFG to the 10K INPUT 2 of the controller, select the sawtooth shape and adjust the amplitude and frequency to be the same as that from the synthesizer. You may need to switch the NOR/INV switch of the DFG to match the synthesizer phase. Then you may switch back and forth between the two inputs with the controller, using the GAIN controls to get the loudness the same for both sources.

SQUARE WAVE:

PRECAUTION: Before proceeding to further experiments using the Fourier synthesizer, or doing any rewiring of the system, turn the system off, following the instructions at the end of this guide.

In the section above on the sawtooth wave, we used the results of mathematical analysis to find the amplitude and phase of the harmonics. In this section, we will directly measure the harmonic content of a square wave using the waveform analyzer, and then build a square wave on the synthesizer using the results of the analysis.

Connect the 10K OUTPUT from generator 1 of the Dual Function Generator (DFG) to the INPUT of the waveform analyzer. Set the Dual Function Generator (DFG) for a 440 hz square wave, and set the waveform analyzer for BAND PASS at 440 Hz. Across the 10K OUTPUT of the waveform analyzer connect the following instruments in parallel: multimeter set for AC volts, frequency counter, and channel 1 on the oscilloscope. Connect channel 2 on the oscilloscope directly across the 10K OUTPUT of GENERATOR 1 of the DFG. Set the oscilloscope to trigger on channel 2. Switch on the DFG, waveform analyzer, multimeter, frequency counter and oscilloscope.

Since the dial on the waveform analyzer does not have the necessary accuracy, you may use a frequency counter to identify the harmonic being passed. Adjust the AMPLITUDE on the DFG for about 1 volt p-p as seen on channel 2 of the oscilloscope. Now adjust the INPUT Gain of the waveform analyzer so that channel 1 on the oscilloscope shows a sine wave. If the GAIN is turned too high the sine wave will be distorted.

Notice that the sine wave has the same frequency f_o as the square wave. Slowly rotate the frequency control on the waveform analyzer until maximum voltage is read on the meter and seen on the oscilloscope. This voltage, V_o , is the amplitude of the $n = 1$ term in the Fourier series. Also notice that the phase of the sine wave is a positive sine wave. The frequency of the sine Wave, f_o , may be read on the frequency counter, and should be $\approx 440\text{Hz}$.

Now, by changing the FREQUENCY control on the waveform analyzer you may determine the amplitude and phase of the higher harmonics of the square wave. Sweep the frequency band slowly, and always tune for maximum voltage before taking measurements. The harmonics of frequency greater than 3960 Hz cannot be duplicated on the synthesizer, but are present and you may be able to find a few of them before they are too faint for detection.

Note that frequency counters are often very touchy in the way they "trigger" on uncertain waveforms and are not altogether easy to use. Often it is easier to identify which harmonic is being observed by using both traces on the oscilloscope simultaneously, feeding the output of the waveform analyzer into one input of the oscilloscope (channel 1) and the input signal to the analyzer into the other input (channel 2). The harmonic is identified by counting how many cycles of output signal fit into one cycle of input signal. *A hint: not all harmonics are present in a square wave.*

Once you have finished your analysis of the square wave, you may use your results to build a square wave on the synthesizer just as you did for the sawtooth wave. Any of the other wave shapes available on the DFG (or in fact any wave shape) may be synthesized in just this manner. Experiment 4 of this set attempts to synthesize some musical instruments.

PHASE SHIFT AND SQUARE WAVE RESPONSE: (an optional section)

PRECAUTION: Do not disturb the settings for the synthesized square wave from the previous section, but do turn off the system, as outlined at the end of this guide, before proceeding to this or further experiments.

A major stereo manufacturer has for some years been using the response of their amplifiers to square waves as a major test of their quality. In particular, the ability to handle 50Hz square waves has been taken to be a good test of the amplifier's ability to handle very low (< 20Hz) frequencies. In this section we investigate this test procedure.

Wire the 10K OUTPUT from the SUMMING AMPLIFIER of the Fourier synthesizer to the INPUT of the waveform analyzer. Set the waveform analyzer to HIGH PASS at a frequency ~100 Hz. Also wire in parallel the 10K INPUT 1 of the controller.

Now wire the 10K OUTPUT from the waveform analyzer to the 10K INPUT 2 of the controller, and to channel 2 of the oscilloscope. Wire the amplifier to the controller and turn on all components. Channel 1 of the oscilloscope shows the synthesized square wave, as in the previous section. Channel 2 displays the square wave after passing through the HIGH PASS filter which removes frequency components below the dial setting of ~100 Hz. Notice the effect of changing the frequency of the cutoff.

You may verify that the percentage tilt P of the square wave is:

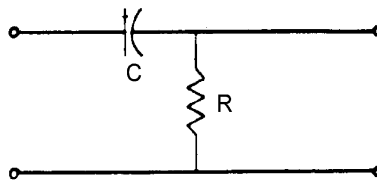
$$P \approx 100\pi \frac{f_c}{f}$$

where: f = square wave frequency and f_c = low frequency cutoff. This result may surprise you.

After all, the square wave is just a sum of sine waves of frequencies 440 Hz, 1320 Hz, 2200 Hz, 3080 Hz etc, and the filter is only taking out frequencies below 100 Hz. A clue to what's going on can be found by adjusting the phase angle of the fundamental of the square wave until the output of the wave analyzer is a square wave again. Of course, the input, as seen on channel 1, will then not be square.

Thus, the effect of the low cutoff is to affect the phase relationship between the fundamental and its harmonics in the square wave, or any other wave shape. Whether or not these altered phase relationships are audible is a question of psychoacoustics which you may investigate by switching back and forth between the original square wave and the tilted wave on the controller. Input 1 is the original; Input 2 is after filtering. Use the GAIN controls on the controller to get the 2 inputs to equal loudness. This phase shift is a characteristic of all circuits which do not have completely flat frequency response from 0 to ∞ Hz.

The simplest type of High Pass filter circuit is:



This circuit is NOT the circuit of the waveform analyzer. Nonetheless, for this simple circuit, the

cutoff frequency f_c (output down 3 dB) is $f_c = \frac{1}{2\pi RC}$

and the phase shift for an input signal of frequency f is

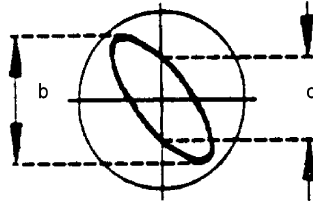
$$\theta = \tan^{-1} \frac{f_c}{f}$$

To investigate the phase response of the waveform analyzer, wire the output of one generator of the DFG to both the INPUT of the waveform analyzer and to channel 2 of the oscilloscope. Wire the 10K OUTPUT of the analyzer to channel 1 of the oscilloscope. Set the DFG for a sine wave of 100 Hz.

First we must set the dial on the analyzer for exactly 100 Hz. To do this switch the analyzer to BAND PASS, and then adjust the FREQUENCY of the analyzer for maximum output. Now, switch the analyzer to HIGH PASS. The phase difference between the input and output signals of the analyzer can be measured in either of two ways. In the first method you can simultaneously display both signals (from channel 1 and channel 2) on the scope in the time-base mode and you can then measure the phase difference on the screen. The second method uses Lissajous figures.

Wire the 10 K OUTPUT from the analyzer to the x input of the oscilloscope with the oscilloscope set in the x - y mode. The analyzer input should be still connected to the y input (channel 2) of the oscilloscope.

The oscilloscope display should look similar to the figure below.



The input signal is displayed vertically simultaneously with the filtered signal displayed horizontally.

The phase angle θ between the two is: $\theta = \sin^{-1} \frac{c}{b}$

You may measure θ for various values of the ratio $\frac{f_c}{f}$ by increasing the input frequency f on the

DFG. How does the phase shift for the waveform analyzer compare to the shift for the simple RC filter discussed above?

TURNING THE SYSTEM OFF:

Follow these instructions in order:

- 1) Turn all GAIN, AMPLITUDE and VOLUME controls to low.
- 2) Turn off all power switches.

EXPERIMENT 4: - HARMONIC ANALYSIS AND SYNTHESIS

INTRODUCTION:

This experiment analyzes musical and other sources for harmonic structure. The results of this analysis are then used to synthesize the same wave pattern on the Fourier synthesizer. The experiment is a direct extension of Experiment 3, which is a prerequisite.

In this experiment we will not go into digital synthesizers. Current digital techniques make it convenient to Fourier analyze signals and to convert that analysis into digital information.

The converse is also performed in which digital signals can then be synthesized into Fourier components of output signals. Thus digitally stored and generated information becomes converted into appropriate sound. You will not be exploring this technology.

APPARATUS:

Fourier Synthesizer	Waveform Analyzer
Controller	Amplifier
Speaker	Oscilloscope
Frequency Counter	Multimeter
Tape for Experiment 4	Tape Player

SETTING UP:

Set the controls of the Fourier synthesizer as follows:

- 1) All AMPLITUDE and GAIN knobs turned down
- 2) All SUMMING AMPLIFIER switches OUT
- 3) Power OFF

Set the controls of the controller as follows:

- 1) Switch position to no. 1
- 2) TAPE switch to "Tape"
- 3) All 3 GAIN controls fully on
- 4) MASTER VOLUME control fully off

The sound source you will use is on the Experiment 4 tape (available at the **Resource Centre**).

Do not turn the tape recorder on yet.

Wire the 10K OUTPUT from the SUMMING AMPLIFIER of the Fourier synthesizer to the 10K INPUT 1 of the controller. Connect the TRIGGER OUTPUT from the synthesizer to the external trigger input of the oscilloscope. Wire the METER outputs of the controller to the INPUT of the waveform analyzer. Also wire the METER outputs of the controller to channel 2 of the oscilloscope. Set the oscilloscope to trigger on channel 2. Across the 10K OUTPUT of the waveform analyzer connect the following instruments in parallel: The multimeter set for AC volts, a frequency counter, and channel 1 on the oscilloscope. Thus, channel 2 shows the wave which will be analyzed for harmonics with the analyzer, meter, counter and channel 1. Set the waveform analyzer for Band Pass Mode. Now, turn on the amplifier, tape player, waveform analyzer, frequency counter, and oscilloscope. You will not use the Fourier synthesizer until later.

ANALYSIS AND SYNTHESIS:

The contents of the Experiment 4 tape are listed with the tape. Choose a source and locate the desired selection on the tape. Each selection is approximately 10 minutes long and is separated by a few seconds of blank. Now, just as for the square wave in Experiment 3, you may sweep the frequency band with the waveform analyzer to determine the relative amounts and phases of the harmonics of the source.

The tapes have been processed by up to 4 different tape decks, and instruments have been reproduced by synthesizers, so don't expect the fundamental to be exactly 440 Hz for all the sources. If the oscilloscope does not display a fairly stable picture on channel 2, you may need to adjust the trigger control on the oscilloscope.

After you have analyzed a source, you may synthesize it on the Fourier synthesizer, and see how good a representation you may achieve. Remember to use the External Trigger on the oscilloscope when using the synthesizer, and to set the TAPE switch to "dual function generator" on the controller. Also remember to touch the RESET button on the synthesizer before adding a harmonic.

TURNING THE SYSTEM OFF:

Follow these instructions in order:

- 1) Turn all GAIN, AMPLITUDE and VOLUME controls to low
- 2) Turn off all power switches

EXPERIMENT 5: - LOUDSPEAKER PRINCIPLES

INTRODUCTION:

In this experiment some of the parameters of a loudspeaker are investigated. Some of the topics investigated include: impedance, dispersion, transient response and frequency response. We expect you to have spent a few hours doing Experiment 1 and/or Experiment 2 from this set before attempting any of these measurements.

You should be aware that loudspeaker design is more art than science, and that any measurements will give only a partial picture of how good, or bad, the speaker sounds when reproducing music. You are invited to bring in your own loudspeaker to test if you wish. Of course, we are not responsible for damaged speakers.

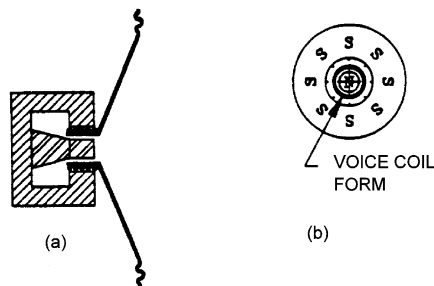
APPARATUS:

Random Noise Tape
Controller
Speaker
2 Multimeters
Precision Sound Level Meter
Tape Player

Dual Function Generator
Amplifier
Oscilloscope
Dispersion Template
Frequency Counter
Microphone

BASIC PRINCIPLES:

A loudspeaker is a device for converting electrical energy into acoustic energy. The most common type is the dynamic paper cone type shown in the figure.



(a) Essential parts

(b) Endview of magnetic structure

Typical Dynamic Loudspeaker

A rigid cone of paper, or less commonly plastic or metal, is held in place by an outer suspension, usually of rubber. The voice coil is wound around the base of the cone, and the interaction of the current in the voice coil with the field the permanent magnet causes the cone to move. An alternating current from the amplifier of frequency f will cause the cone to oscillate with the same frequency f . This, in turn, generates a sound wave also of frequency f .

Since the loudspeaker is a mechanical vibrating system, it has a resonance frequency, and one popular way of controlling this frequency is to place the speaker in a totally sealed box of another resonance frequency. In this "acoustic suspension" type of system, the air inside the box forms a cushion to control the cone's movements; thus the suspension itself only holds the cone in place around the magnet. The combined resonance frequency of speaker plus enclosure in this type of system is usually the lower limit of bass response.

There are, of course, loudspeakers based on other principles. The "electrostatic" speaker is essentially a big capacitor whose plates vibrate if an alternating voltage is placed across them. The "piezoelectric" driver utilizes the fact that certain materials respond to voltages by generating mechanical forces, and vice-versa. Piezoelectrics are also used in some phonograph cartridges and microphones.

Victor Campos, head of loudspeaker design at KLH, is fond of saying that the three main parameters of a speaker are efficiency, bandwidth and low distortion, and that it is impossible to get all three at once. Efficiency is the amount of sound energy produced for a given input of electrical energy; thus the efficiency of a speaker determines only how much amplifier power is needed to drive it. Bandwidth is how wide a frequency range the driver can reproduce; thus the bandwidth of the drivers determines how many different speakers in a system are needed to cover the entire range of audible sound. Distortion is a measure of how well the output acoustic wave shape matches the input electrical signal.

IMPEDANCE:

One of the parameters of loudspeaker is its impedance Z . Just as the resistance R is used in Ohm's law for DC circuits:

$$V = IR \quad (\text{DC circuits})$$

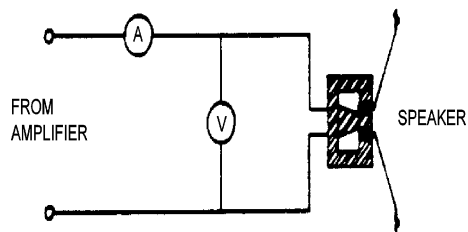
the impedance Z is used in Ohm's law for AC circuits:

$$V = IZ \quad (\text{AC circuits})$$

The big difference is that the resistance is usually a constant number, while the impedance depends on the frequency of the AC. For a complicated circuit such as a speaker system with coils capacitors, mechanical resonances etc., the behaviour of Z with frequency f is easier to measure than it is to derive.

In order to measure the impedance of the loudspeaker, set up the apparatus just as in Experiment 1 of this set, but **do not turn on amplifier yet**, and **do not turn the DFG on**.

Now look at the back of the loudspeaker. The cord from the amplifier terminates in 2 banana plugs which may be unplugged from the speaker so an AC ammeter and AC voltmeter may be inserted according to the figure.



Make sure the leads are not shorted, and then plug in the amplifier power cord and turn the DFG on. Adjust the DFG for a 300Hz sine wave, and turn up the MASTER VOLUME control to a moderate level (100mV or so as measured at the speaker), and measure the current and voltage for a number of different frequencies without changing any of the GAIN or VOLUME controls. Pay particular attention to the impedance at resonance, which is around 40Hz in our speaker, and at crossover, which is around 1.8kHz in our speaker.

A good way to present your data is to plot Z versus f on semi-log graph paper, with frequency plotted on the log axis. A few moments thought on the relationship between frequency and octaves should make it clear why this is a good technique.

You may wish to ponder the fact that the loudspeaker's output is approximately independent of frequency in spite of the large variations in impedance.

DISPERSION

Since sound is a wave, interference and diffraction effects can occur. Most dynamic loudspeakers put out bass notes equally in all directions ("omnidirectional") precisely because the wavelength λ is large compared to the size of the driver, so diffraction is not a problem (the wavelength of a 300Hz tone is over one meter). Thus, a low frequency driver may be rather large. However at higher frequencies the driver tends to "beam", and you may have to be directly in front of the speaker to hear the cymbals (the wavelength of a 5kHz tone is about 7cm). Thus, a high frequency driver is usually small. Because the speaker is not a perfect plane wave source, it is easier to measure these directional characteristics than to derive them.

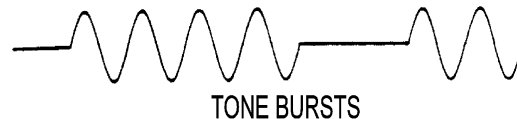
Set up the equipment as in the previous section on impedance. The 2 multimeters from that section are not needed here (available at the Resource Centre); but **be sure to turn off amplifier before changing any of the wiring at the back of the speaker!** Plug the microphone (available at the Resource Centre) into channel 2 of the oscilloscope, and position it 50cm to 1m away from the speaker. There is already prepared a large piece of cardboard with degree markings on it (available at the Resource Centre). Measure the output of the speaker *versus* angle for a few different frequencies (perhaps 300Hz, 5kHz, and 9kHz). Plot your results, for each frequency, in terms of decibels on polar graph paper.

TRANSIENT RESPONSE

The cone of a dynamic loudspeaker must be strong so it does not flex when being driven by the voice coil, yet also light so it responds quickly to the impulse. Also, although it is an oscillating system it must stop when no voltage is applied to the coil. In this section the ability of the speaker to do this is investigated by means of tone bursts.

Set up as in the previous section: *i.e.*, as in part 1 of this experiment, with the microphone mounted on a stand 50cm or so from the speaker and plugged into channel 2 of the oscilloscope. On the DFG, switch generator 2 in the summing amplifier, adjust for a 300Hz sine wave. Set generator 1 for a square wave of 30 to 50Hz, and switch the lower MODULATION switch to GATE. Trigger the oscilloscope on the TRIG OUTPUT from generator 1.

Channel 1 on the oscilloscope shows the input to the speaker, and should resemble the figure.



Channel 2 shows the loudspeaker output. Try a number of different frequencies for generator 2, and see what conclusions you may draw about the speaker's performance. Polaroid photographs of the oscilloscope display your data well.

FREQUENCY RESPONSE

The frequency response of a loudspeaker is one of its most important parameters. Unfortunately, it is also one of the most difficult properties to measure. If the loudspeaker is driven by a pure sine wave, two problems arise. First, a pure sine wave is rarely encountered in music or speech. Thus, the loudspeaker's response to sine waves is not truly indicative of its normal performance.

Second, and more important, a pure sine wave will excite standing waves in the room where the speaker is being tested, and further these standing waves are more severe for a pure sine wave than for the more transient signals of music. Years ago, Acoustic Research tested their speakers outdoors to attempt to overcome this problem. A good multi-kilo-dollar anechoic chamber helps to reduce, but not eliminate, this problem.

Thus, usually one drives the loudspeaker with a random noise source, something similar to the hiss between stations on FM radio. This reduces the effect of environment on the results. There are two standard types of random noise. "White noise" is random noise with constant frequency response across the audible range (*i.e.*, constant power per Hz over the full frequency range). However, this type of noise means the octave from 20 to 40Hz has only $\frac{1}{2}$ the amount of signal as the octave from 40 to 80Hz. Since the octave is only $\frac{1}{2}$ as wide when measured in frequency units (Hz).

The other type of noise, called $\frac{1}{f}$ (one-over-f) or "pink noise". $\frac{1}{f}$ noise ends up having equal noise power for every octave in the sound spectrum, no matter what frequency is being considered.

(You might think about what happens to the power spectrum of white noise as $f \rightarrow \infty$ or to the power spectrum of $\frac{1}{f}$ noise as $f \rightarrow 0$.)

There are still environment factors when using a random noise source, which is a reflection of the fact that the same loudspeaker sounds quite different in different living rooms, or in different positions in the same room. However, doing the test in an anechoic chamber at least provides a standard, if non-typical, environment.

Finally, near a cross-over point two or more drivers can be simultaneously reproducing a tone, and interference effects dependent on microphone position can occur. This is not a severe problem in a living room because over 60% of what reaches your ear has been reflected off the walls, floor and ceiling, and this tends to cancel these effects.

Bear in mind in interpreting your results and these graphs that octave filters give only the average output for the entire band. For example, the lowest band gives from 23Hz to 47Hz, although there are not more than half a dozen speakers in the world that will accurately produce a 23Hz tone.

For a random noise source, we have a Random Noise Tape prepared (available at the **Resource Centre**).

The Precision Sound Level Meter (available at the **Resource Centre**), with attached Octave Filter Set, and a Type 4145 microphone (available at the **Resource Centre**) will be used to measure the output of the loudspeaker. Use the extension cable A0-0033 to connect the Sound Meter to the microphone mounted on the stand, positioned 50cm to 1m away from the loudspeaker, and pointed towards it. There is no need to use the Random Incidence Corrector for this measurement, nor is there any need to calibrate the meter since only relative output is of interest here. Now you may "sweep" the frequency band with the Octave Filter. Finally, the microphone type 4145 only has a response up to 18kHz, although the 16kHz band on the filter set extends to 24kHz. Thus a correction must be added to your result for the highest band. We leave as an exercise for you to show that the correction should be about 3dB.

TURNING THE SYSTEM OFF:

Follow these instructions in order:

- 1) Turn all GAIN, AMPLITUDE and VOLUME controls to low.
- 2) Turn off all power switches.

EXPERIMENT 6: - SOUND LEVELS

The Precision Sound Level Meter (available at the **R**esource **C**entre) may be used to measure sound levels in the environment (Room 126, St. George Street, etc.) full operating instructions are contained in the Tech. Sheet on the meter. Usually sound levels are done with A weighting, but you may also measure unweighed sound levels (linear), and do spectral analysis with the octave filter set. Be sure you understand the concepts discussed in *Experiment 1 - Frequency, Pitch and Decibels* before attempting to use the meter. Also, the meter is of professional quality, and **must** be treated with care. A chart of sound levels is posted in room 125 which may help you interpret your results.

(dh - 77, jbv - 89, cp - 93)