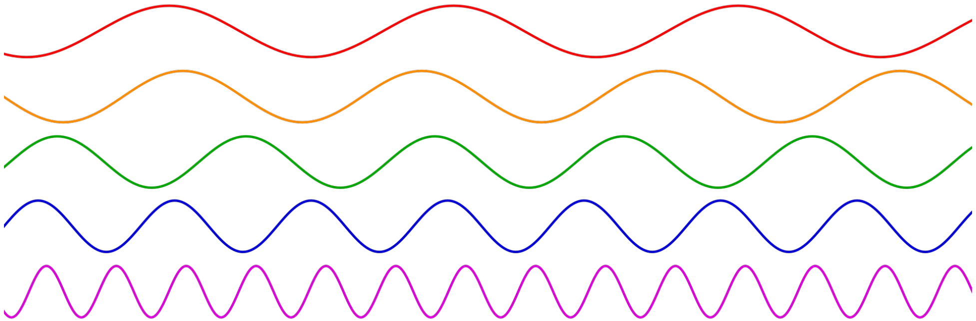
Sound

Background Information - Compression Waves

- Sound energy from a source to an observer of the sound
- In sinusoidal form
- **Peak** - where air is compressed in the wave
- **Trough** - where is air is decompressed
- Waves are parallel to direction of travel



Background Information - Generation of Sound

- *Hearing* a sound consists of:
    - Auditory nerves being stimulated to pass signals to the auditory centers of the brain
    - Stimulation occurs via vibrations transmitted over some medium
        - Medium is usually air
        - Vibrations is a compression wave
- If a tree falls in a forest and no one is there to hear it fall does it make a sound?
    - Yes ... and no!

Characteristics of Sound - Frequency/Period

- Frequency
  - Unit of measure is a hertz (Hz)
  - Number of occurrences of a *repeating* event per unit of time
    - Hertz is repetitions per second
- Period
  - Inverse of frequency
    - 1/Hertz
  - Length of time of 1 iteration of an event

Characteristics of Sound - Amplitude

- The size of the wave
  - i.e. its magnitude
- The maximum extend of a vibration for oscillation
  - Measured from the position of equilibrium to either peak or trough
  - Measure by taking distance from peak to trough and divide by 2

Characteristics of Sound - Phase

- The shift of the starting position of the vibration (compression wave)
- Not (normally) an audible characteristic of sound
- Important in the interactions between different waves
- Example:
  - sinΘ and cosΘ have the same wave form
  - cosΘ phase is +1 to sinΘ since it has the value 0 at 90°
- https://cmtext.indiana.edu/acoustics/chapter1_phase.php

Characteristics of Sound - Speed

- Not relevant for use in the course but ...
- Sound takes time to travel
  - Example: thunder heard after lightning
- Calculated using Newton-Laplace equation
  - $c = \sqrt{K/\rho}$
- Speed is different in different mediums
  - Sound travels faster in water than air

Analog Audio Signals and Transducers

- Transducer converts energy from one form to another
  - Speakers converts voltage to sound waves
  - Microphone converts sound wave to voltage

Tone & Music

- Tone is:
  - A particular frequency that can be heard
  - A named frequency
- Music is:
  - A combination of tones
  - Each tone is *played* for a set duration
  - Duration for each tone can vary

Sound and Computers

- Computers must store and process audio signals digitally.
- Pulse-code modulation (PCM) is a digital representation of an analog signal
    - Achieved by repeatedly “sampling” and “quantizing” the input signal
- A way of representing an analog signal as a set of distinct digital values (states)
    - Done via repeated sampling.

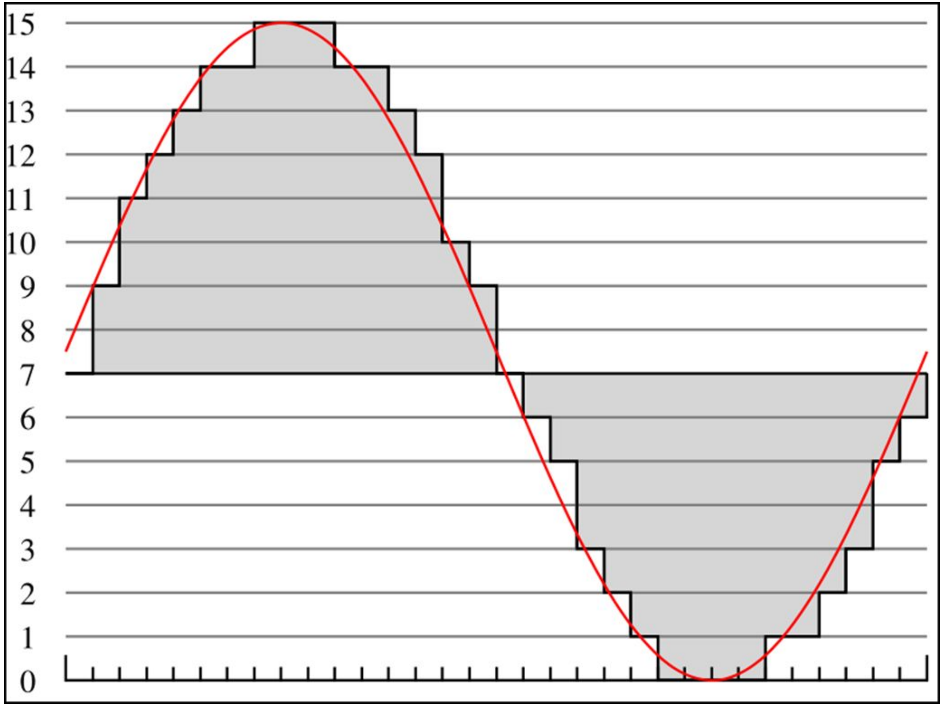
PCM

- The amplitude of the analog signal is sampled at uniform intervals
    - Each sample is quantized to the nearest value within a range of digital steps
- Example:
    - Consider a perfect 1 Hz sine wave signal. The following diagram shows the analog signal superimposed on the corresponding digital signal, assuming a sample rate of 32 Hz and a quantization bit depth (# of data bits) of 4 (the quantization function is the ceiling function):

PCM

souce: Wikipedia

- Vertical axis is signal value
- Horizontal axis is time
- Digital signal is a stream of values between 0 and 15
    - Signal can be represented in a 4-bit (nibble) value
- Signal is: 1001, 1011, 1100, ...



PCM - Headroom

- Headroom is the difference between the top of the wave and the top increment
- In example the headroom is zero
  - Maximum value is equal to the top increment
- What if the headroom was not zero? (i.e. sine wave ends at 12)
  - This is a loss of fidelity, since not all bits are used
  - Fidelity is the degree of exactness in the sound reproduction

PCM

- Is the PCM signal a faithful reproduction of the analog signal?
  - NO
- How can we increase the fidelity?
  - Increase the sample rate
  - Increase the quantization bit depth

PCM

- Note:
  - Quantization error (a form of distortion, similar to analog noise)
    - The difference between the analog signal and the closest available digital value at each sampling instant.
  - Frequencies above ½ the sample rate will be distorted or lost
    - In this case at 32Hz – any sound above 16Hz is lost

PCM - Encoding Examples

- CDs encode sound as PCM data. CD-quality audio is 16-bit @ 44.1 kHz per channel
  - There are two channels (stereo)
  - The noise-to-signal ratio from quantization error is (supposedly) below the threshold of human hearing (signal / noise = dB)
  - Most humans cannot (supposedly) hear frequencies above 22 kHz

PCM - Encoding Examples

- DVD quality audio is much higher (up to 24-bit @ 192 kHz), but the raw data is not stored in PCM format
  - It is compressed
  - There can also be more than two channels
  - Example: 5.1 Surround

PCM

- An analog-to-digital converter (ADC) is a circuit that converts an input analog signal to an output PCM signal
- A digital-to-analog converter (DAC) performs the reverse
- PCM is a standard audio format for computers and digital communication
  - The data may actually stored in a compressed format
- E.g. WAV files, although multi-format, usually contain uncompressed PCM data (plus header metadata)
- There are various compression technologies, some of which are “lossy” (e.g. MP3)

Sound Cards

- A sound "card" may or may not be an actual card. It may be:
  - an expansion card, such as a PCI card
  - built in to the motherboard
  - an outboard USB device
  - etc.

Sound Cards

- A sound card includes at least one sound chip, which includes a DAC
  - and maybe an ADC
- At the very least, the card produces an analog audio signal from digital data such as PCM data
  - The audio signal is output on a connector such as a headphone jack.

Sound Cards - Ports/Connectors

- Typical ports/connectors include:
  - line out ← standard 1/8” connector, maybe to headphones or amp
      - (active monitors have a built-in amp)
  - line in ← if input supported
  - mic in ← similar to line in, but for lower level signals needing amplification

Sound Cards - Ports/Connectors

- Typical ports/connectors include:
  - S/PDIF port ← for connecting digital components digitally (Sony/Philips Digital Interconnect Format)
  - MIDI port ← musical instrument digital interface – a protocol for communicating musical note/event data between electronic instruments (but not audio data)
- The number of input/output channels varies. On consumer cards, 0-1 input channels and 1-2 output channels (mono/stereo) is common. In professional settings the numbers are higher

Sound Cards - Typical Functions

- Typical functions include:
  - playback and recording of PCM data (and other formats)
  - sound synthesis
  - mixing of multiple audio “voices”
  - effects (e.g. filtering, reverb, ...)
  - recording of MIDI events; control of MIDI devices

Sound Cards - Playing Sound

- The outbound data must often be double buffered
  - The sound card is playing data in a current buffer
  - The next buffer of data is being prepared (uncompressed)
    - The next buffer must be fully ready before the current buffer is exhausted; otherwise, a drop out may be noticeable
- Increasing buffer size:
  - Decreases the chance of a drop out
  - Increases latency (time delay for conversion)

Atari PSG Overview

- A **P**rogrammable **S**ound **G**enerator (PSG) is a sound chip which produces analog audio signals
- The Atari ST's PSG, the YM2149, is typical of 1980's PSGs found in home computers and arcade games
- In short, this chip produces square wave audio signals by dividing the frequency of a clock signal
  - The wave's frequency and amplitude can be varied by altering the contents of onboard registers

Atari PSG Overview

- The chip doesn't play PCM data (well, actually ...)
- The YM2149 has 16 CPU-accessible 8-bit registers.
- It has three separate channels: A, B and C. Each has:
  - 12-bit tuning (spread across a fine and coarse tune register)
  - A volume register
- Each of channels A, B, and C can be enabled/disabled independently
- There is also a 'Noise' channel
  - 5-bit tuning on a single register

Atari PSG Overview

- The YM2149 has three separate output pins for A, B and C
  - It doesn't mix them itself
  - On the Atari ST, additional hardware mixes them together and outputs a mono signal on the video port's "audio out" pin (the speaker was housed in the monitor)
- The chip also features a noise generator (also square wave, but frequency modulated) and an envelope generator
  - These can be used for sound effects (e.g. an explosion)

Atari PSG Overview

- All 16 registers are mapped through two addresses!
  - One address gives access to the “register select” register.
  - The PSG makes the selected register available at the second address
  - This saves register select pins.
- Strangely, the YM2149 also supports two 8-bit parallel ports

Atari PSG Register Breakdown

The contents of the register array are shown in Table 3.

Register		B7	B6	B5	B4	B3	B2	B1	B0
R0	Frequency of channel A	8 bit fine tone adjustment							
R1						4 bit rough tone adjustment			
R2	Frequency of channel B	8 bit fine tone adjustment							
R3						4 bit rough tone adjustment			
R4	Frequency of channel C	8 bit fine tone adjustment							
R5						4 bit rough tone adjustment			
R6	Frequency of noise				5 bit noise frequency				
R7	I/O port and mixer Settings	I/O		Noise			Tone		
		IOB	IOA	C	B	A	C	B	A
R8	Level of channel A				M	L3	L2	L1	L0
R9	Level of channel B				M	L3	L2	L1	L0
RA	Level of channel C				M	L3	L2	L1	L0
RB	Frequency of envelope	8 bit fine adjustment							
RC		8 bit rough adjustment							
RD	Shape of envelope					CONT	ATT	ALT	HOLD
RE	Data of I/O port A	8 bit data							
RF	Data of I/O port B	8 bit data							

(Register numbers are indicated in hexadecimal notation)
Table 3 Register Array

Rough tone adjustment register	Channel	Fine tone adjustment register
R1	A	R0
R3	B	R2
R5	C	R4