EBU5303

Multimedia Fundamentals

Digital Video and Audio

Agenda

- A video is a sequence of images
- A sound is characterised by its frequency (pitch) and amplitude (loudness)
- CD standard quality is 44,100 Hz (sampling) and 16 bits (quantisation)
- Speech signals contain 3 types of sound, some of them are used for speech recognition
- MIDI format for music stores information such as instrument specification, beginning and end of a note, basic frequency, etc.

Reading

http://burg.cs.wfu.edu/TheScienceOfDigitalMedia/Chapter4/Ch4ScienceOfDigitalMedia.pdf

4.2 Audio Waveforms

4.4 Sampling Rate and Aliasing

4.5.1 Decibels and Dynamic Range

4.6.1 Time and Frequency Domains

4.8 MIDI

http://digitalsoundandmusic.com/

Learning Objectives (1)

- Understand how waveforms represent changing air pressure caused by sound.
- Be able to apply the Nyquist theorem to an understanding of digital audio aliasing.
- Given a sampling rate, be able to compute the Nyquist frequency.
- Given the frequency of an actual sound wave, be able to compute the Nyquist rate.
- Understand the relationship between quantisation level (i.e. sample size or bit depth) to dynamic range of an audio file.
- Given air pressure amplitude for a sound, be able to compute decibels, and vice versa.
- Given a bit depth for digital audio, be able to compute the signal-to-quantisation noise ratio.

Learning Objectives (2)

- Understand the application of the Fourier transform for digital audio processing.
- Be able to read and interpret a histogram of a waveform.
- Understand the information provided in a frequency or spectral analysis of an audio wave.
- Understand the difference between the formats of MIDI and digital audio sound files.
- Become familiar with basic terminology of MIDI and related areas in musical acoustics and musical notation.

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Video - definitions

- **Video** is the technology of electronically capturing, recording, processing, storing, transmitting, and reconstructing a sequence of still images representing scenes in motion.
- **Frame rate**: the number of still pictures per unit of time of video.

- **Analog video**: video recording method that stores continuous waves of red, green and blue intensities.
- **Digital video**: video recording system that works by using a digital rather than an analog video signal.

Refresh rate and frame rate

- The **refresh rate** is the number of times in a second that the display hardware draws the data (i.e. repeated drawing of identical frames).
- The **frame rate** measures how often a video source can feed an entire frame of new data to a display.
- Typical rates: 24 frames per second (frame rate) and 48 or 72 Hz (refresh rate).

Frame Rates

Video Type	Frames Per Second (fps)
NTSC	29.97
PAL	25
SECAM	25
Motion Picture Film	24

NTSC was 30 fps for black-and-white TV, Frame rate was lowered to 29.97 fps to accommodate for color encoding.

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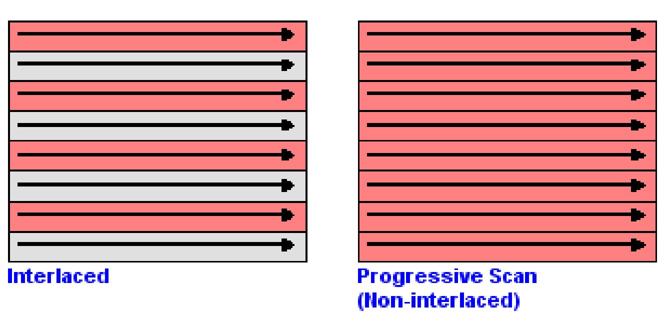
Interlaced vs Progressive

- **Interlaced** scanning displays alternating sets of lines. Because each field happens so quickly we are given the illusion of a whole image.
- **Progressive** video displays the entire image.



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Exercise

A 30fps digital video uses 352 by 255 pixels video frames with a pixel depth of 8.

i) Calculate the size of 1 second of data.

ii) What compression ratio would be needed to transmit 1 second of data in real-time over a 64 Kbps communication channel?

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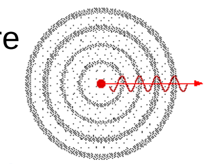
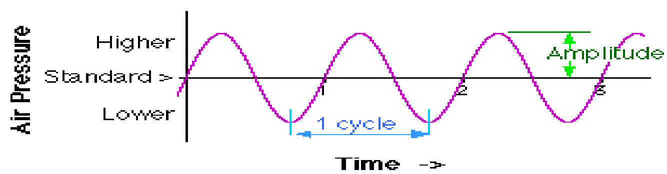
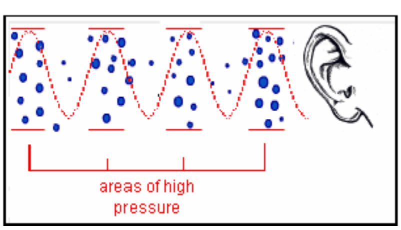
Sound

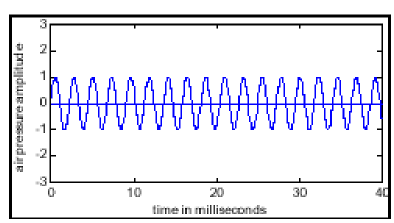
- Sound is a physical phenomenon produced by the vibration of matter, such as a violin string, or a block of wood.
- As the matter vibrates, pressure variations are created in the air surrounding it.
- This alteration of high and low pressure is propagated through the air in a wave-like motion.



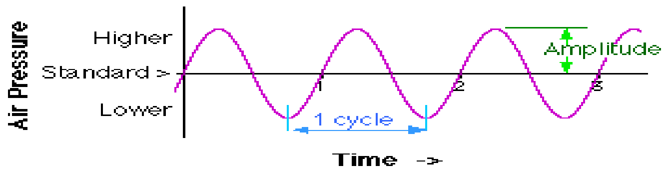


Sound in the analogue domain





Characteristics of Sound Waveforms



- Frequency determines the pitch
  (higher frequency = higher pitch)
  - Infra-sound: from 0 to 20 Hz
  - Human hearing frequency range: 20 Hz – 20 kHz
  - Ultrasound: from 20 kHz to 1 GHz
- Amplitude of the wave determines the volume or intensity (a property subjectively heard as loudness).

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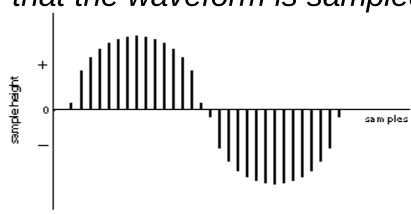
Computer Representation of Sound - Sampling -

- A computer measures the amplitude of the waveform at regular time intervals to produce a series of number (sampling). This is done by an ADC (*Analog-to-Digital Converter*)
- Sampling rate: the rate at which a waveform is sampled. e.g. *the CD standard sampling rate of 44100 Hz means that the waveform is sampled 44100 times / second*.

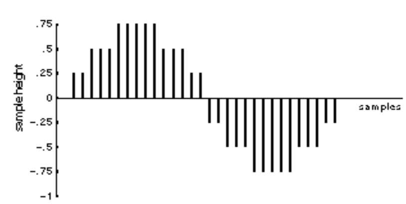


Sampled waveform

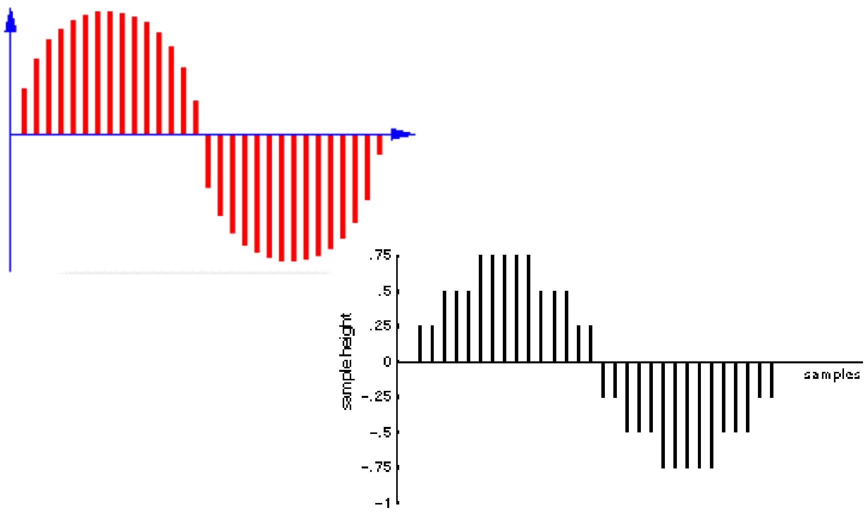
Computer Representation of Sound - Quantization -

- Quantization: the resolution or quantization of a sample value depends on the number of bits used in measuring the height of the waveform (*usually 8-bit or 16-bit*)

3-bit quantization



Digitisation of Sound



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Exercise

A high-quality (CD standard at 44.1KHz) audio signal with 2 channels of 16-bit samples is transmitted uncompressed over an ISDN 64Kbits/s communications channel.

i) Calculate the number of seconds taken to transmit a one-second burst of audio

ii) Estimate what compression ratio would be needed to transmit the audio in real-time.

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Reminder: Nyquist theorem

Sample twice as often as the highest frequency you want to capture

Let f be the frequency of a sine wave. Let r be the minimum sampling rate that can be used in the digitisation process such that the resulting digitised wave is not aliased. Then:

$$r = 2f$$

r is called the ***Nyquist rate***.

Nyquist Rate and Nyquist Frequency

- Given an actual frequency to be sampled, the ***Nyquist rate*** is the lowest sampling rate that will permit accurate reconstruction of an analog digital signal.
- Given a sampling rate, the ***Nyquist frequency*** is the highest actual frequency component that can be sampled at the given rate without aliasing.
- Based on the Nyquist theorem, the Nyquist frequency is half the given sampling rate.

Nyquist Rate and Nyquist Frequency

KEY EQUATION

Given $f_{\max}$, the frequency of the highest-frequency component in an audio signal to be sampled, then the ***Nyquist rate***, f_{nr}, is defined as

$$f_{nr} = 2f_{\max}$$

KEY EQUATION

Given a sampling frequency f_{samp} to be used to sample an audio signal, then the ***Nyquist frequency***, f_{nf}, is defined as

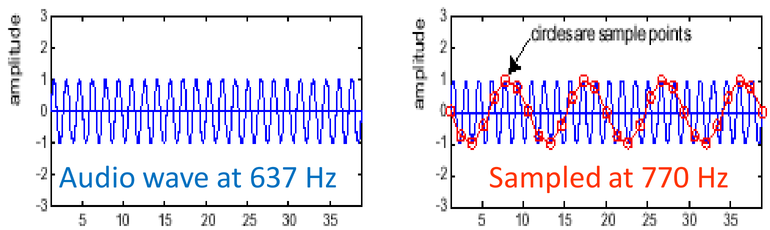
$$f_{nf} = \frac{1}{2}f_{samp}$$

Exercise

The bandwidth of a music signal is between 15 Hz and 20 KHz, assuming the Nyquist sampling rate is used, with 16 bits per sample:

- Derive the bit rate that is generated by the digitisation procedure
- What is the memory in Mbytes required to store a 10 minute passage of stereophonic music?

Aliasing (sampling error)



- The reason a too-low sampling rate results in aliasing is that there aren’t enough sample points from which to accurately interpolate the sinusoidal form of the original wave.
- If we take ***more*** than two samples per cycle on an analog wave, the wave can be precisely reconstructed from the samples.

Measuring Sound Amplitude in Decibels

- A decibel is not an absolute unit of measurement.
- A decibel is always based upon some agreed-upon reference point, and the reference point varies according to the phenomenon being measured.
- For sound, the reference point is the *air pressure amplitude for the threshold of hearing*.
- A decibel in the context of sound pressure level is called ***decibels-sound-pressure-level*** (***dB_SPL***).

Measuring Sound Amplitude in Decibels

KEY EQUATION

Let E be the pressure amplitude of the sound being measured and E_0 be the sound pressure level of the threshold of hearing. Then ***decibels-sound-pressure-level***, (***dB_SPL***) is defined as

$$dB\_SPL = 20 \log_{10}\left(\frac{E}{E_0}\right)$$

E_0 = 0.00002 Pa

Exercise

- What would be the amplitude (in decibels) of the audio threshold of pain, given as 30 Pa?
- what would be the pressure amplitude of normal conversation, given as 60 dB?

Measuring Sound Amplitude in Decibels

- dB_SPL is an appropriate unit for measuring sound because the values increase logarithmically rather than linearly.
- This is a better match for the way humans perceive sound.
- Experimentally, it has been determined that if you increase the amplitude of an audio recording by 10 dB, it will sound about twice as loud.
- For most humans, a 3 dB change in amplitude is the smallest perceptible change.

Measuring Sound Amplitude in Decibels

Approximate decibel levels of common sounds:

Sound	Decibels (dB_SPL)
Threshold of hearing	0
Rustling leaves	20
Conversation	60–70
Jackhammer	100 (or more)
Threshold of pain	130
Damage to eardrum	160

Signal to Quantisation Noise Ratio (SQNR)

- SQNR is also measured in decibels.
- SQNR is directly related to ***dynamic range:*** the ratio of the largest sound amplitude and the smallest that can be represented with a given bit depth.

Let *n* be the bit depth of a digitised media file (e.g. digital audio). Then the signal-to-quantisation noise ratio ***SQNR (or dynamic range)*** is:

$$SQNR = 20 \log_{10}(2^n) = 20n \log_{10}(2) \sim 6n$$

Dynamic Range

KEY EQUATION

Let n be the bit depth of a digital audio file. Then the ***dynamic range of the audio file***, d, in decibels, is defined as

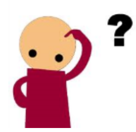
$$d = 20n \log_{10}(2) \approx 6n$$

- You can estimate that an *n*-bit digital audio file has a dynamic range (or, equivalently, a signal-to-noise-ratio) of 6*n* dB.
- Dynamic range is a relative measurement—the relative difference between the loudest and softest parts representable in a digital audio file, as a function of the bit depth.

Exercise

- What is the dynamic range (SQNR) of a 16 bit digital audio file?
- How about a 8 bit digital audio file?

Question



A sound file encoded with a 8 bits quantisation rate is likely to be:

- A piece of music
- Natural sound (e.g. rain)
- Speech
- A song

Quantisation Error

- While an insufficient sampling rate can lead to aliasing, an insufficient bit depth can create quantisation error.
- ***Audio dithering*** is a way to compensate for quantisation error. The way to do this is to add small random values to samples in order to mask quantisation error.
- ***Noise shaping*** is another way to compensate for the quantisation error: it redistributes the quantisation error so that the noise is concentrated in the higher frequencies, where human hearing is less sensitive

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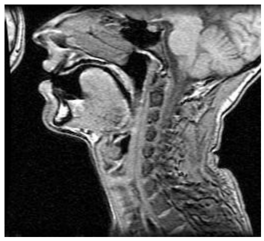
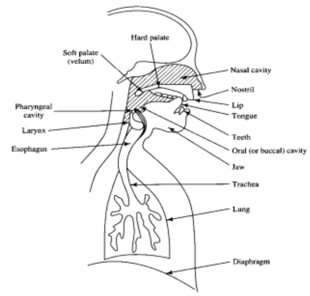
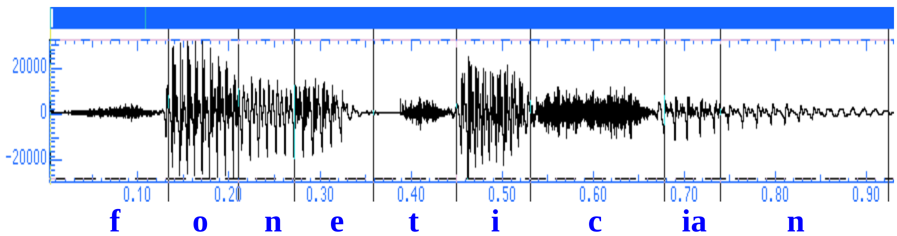
Speech

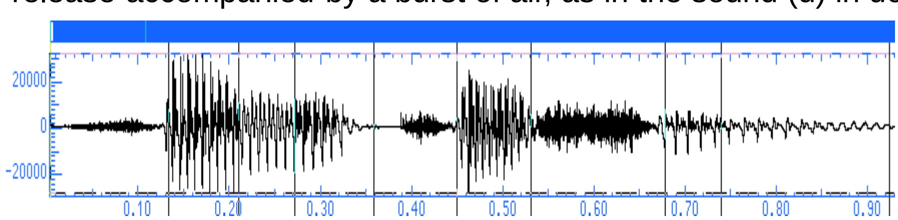






Types of Speech Sounds

- Voiced sounds : the vocal chords are vibrated, which can be felt in the throat. All vowels are voiced.
- Fricatives (unvoiced sounds) : a consonant, such as *f* or *s* in English, produced by the forcing of air through a constricted passage.
- Plosives (also unvoiced sounds) : a speech sound produced by complete closure of the oral passage and subsequent release accompanied by a burst of air, as in the sound (d) in *dog.*



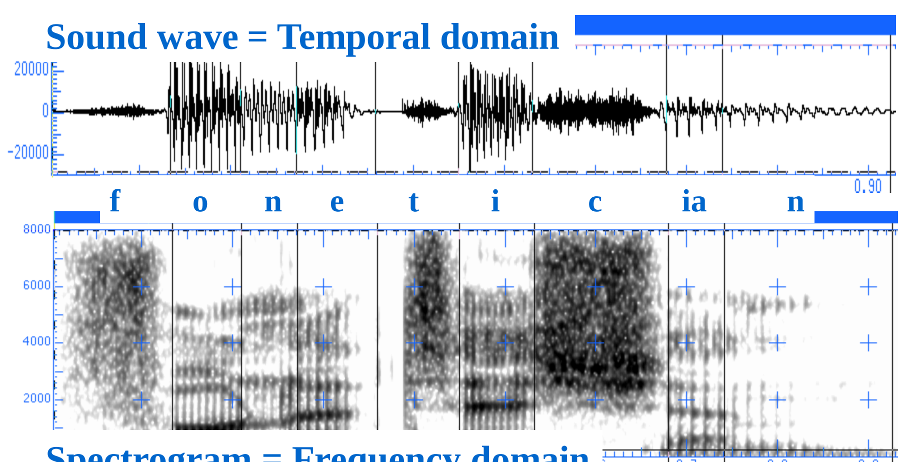
Voiced Speech Sounds

Voiced speech sounds have two properties which can be used in speech processing:

▪ Speech signals show during certain time intervals almost periodic behaviours. These signals are *quasi-stationary signals* for around 30 ms.

▪ The spectrum of speech signals (voiced sounds) shows characteristic maxima. These maxima, called formants, occur because of resonances of the vocal tract.

Temporal and Frequency Domains

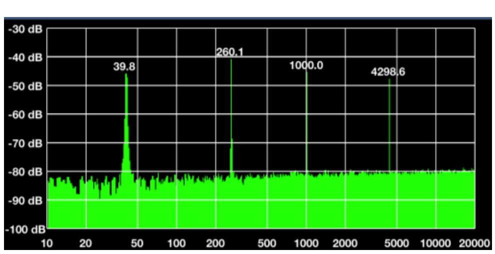


Temporal and Frequency Domains

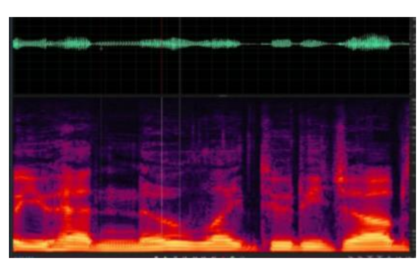
- Sound can be represented either over the time domain or the frequency domain.
- The transform that is used most frequently in digital audio processing is the Fourier transform.
- A complex waveform is equal to an infinite sum of simple sinusoidal waves, beginning with a ***fundamental frequency*** and going through frequencies that are integer multiples of the fundamental frequency.
- These integer multiples are called ***harmonic frequencies***.
- In the ***frequency domain***, data is stored as the amplitudes of frequency components

Temporal and Frequency Domains

The time domain and frequency domain are equivalent. They both fully capture the waveform. They just store the information about the waveform differently.



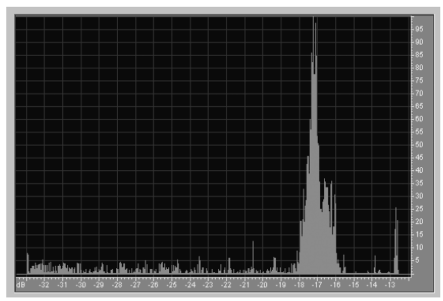
Power Spectrum



Spectrogram

Audio Histogram

- Audio processing programs sometimes offer a statistical analysis of your audio files, which analyse sample values in the **time domain**.
- An ***audio histogram*** shows how many samples there are at each amplitude level in the audio selection.



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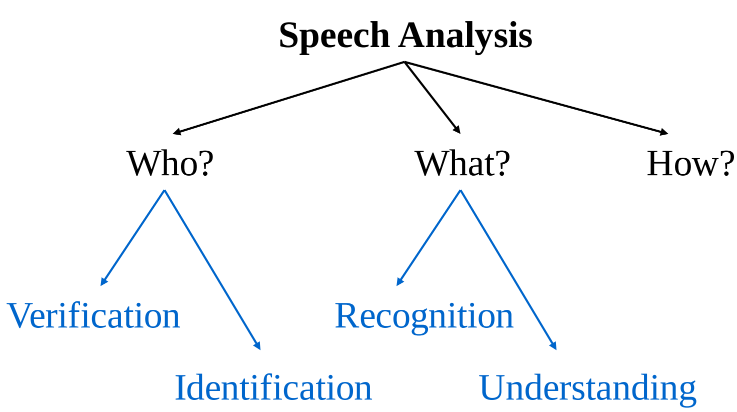
Speech Processing Applications



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Music

• Thus far, we've been considering digital audio that is created from sampling analog sound waves and quantising the sample values.

• There's another way to store sound in digital form: ***MIDI (Musical Instrument Digital Interface)***.

• MIDI stores "sound events" or "human performances of sound" rather than sound itself.

• The difference between sampled digital audio and MIDI is analogous to the difference between bitmapped graphics and vector graphics

MIDI Data Format

• A MIDI file contains messages indicating when a note begins (Note On), when a note ends (Note Off), what the note is, how hard it is pressed (Velocity), how hard it is held down (Aftertouch), what instrument is played, and so forth.

• Each MIDI message communicate one musical event.
e.g. when a musician presses a piano key, the MIDI interface creates a MIDI message where the beginning of the note with its stroke intensity is encoded.

• The MIDI standard identifies 128 instruments (including noise effects) with unique numbers (e.g. 41 for the violin).

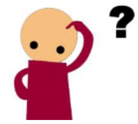
MIDI Hardware and Software

• Hardware devices that generate MIDI messages are called ***MIDI controllers***.

• Devices that read MIDI messages and turn them into audio signals are called ***MIDI synthesizers***. Two methods for synthesizing sound are ***frequency modulation synthesis*** and ***wavetable synthesis***.

• A ***MIDI sequencer*** is a hardware device or software application program that allows you to receive, store, and edit MIDI data.

Questions



- Why are MIDI encoded music signals very small?
- What other advantage to MIDI audio is there compared to sampled digital audio?
- Is there any disadvantage to MIDI audio compared to sampled digital audio?

Musical Acoustics and Notation

• In Western music notation, musical sounds, called ***tones,*** are characterized by their pitch, timbre, and loudness.

• With the addition of onset and duration, a musical sound is called a ***note***.

• The ***pitch*** of a note is how high or low it sounds to the human ear.

• The ***timbre*** of a musical sound is its "tone color".

• The lowest frequency of a given sound produced by a particular instrument is its **fundamental frequency**. Then there are other frequencies combined in the sound, which are integer multiples of the fundamental frequency, referred to as ***harmonics.***

Exercise

- A musical note played on an instrument consists of a fundamental frequency and, depending on the instrument, different numbers of harmonics. Each harmonic is an integer multiple of the fundamental frequency.
- Given a note from a musical instrument, which contains only the following frequency components: 100Hz, 200Hz, 300Hz, and 400Hz, **at what rate would you need to sample this sound to ensure that the sampled audio was of the same fidelity as the original note?**
- Assuming that the amplitude of each harmonic is half the amplitude of the previous harmonic, **sketch the signal in the frequency domain for the above note.**

Musical Acoustics and Notation

If the frequency of one note is 2^n times of the frequency of another, where n is an integer, the two notes sound "the same" to the human ear, except that the first is higher-pitched than the second.

KEY EQUATION

Let g be the frequency of a musical note. Let h be the frequency of a musical tone n ***octaves*** higher than g. Then

$$h = 2^n g$$

Exercise

If the frequency of a note A is about 440 Hz, what is the frequency of an A two octaves below the 440 Hz A?

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