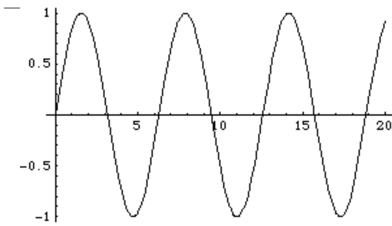
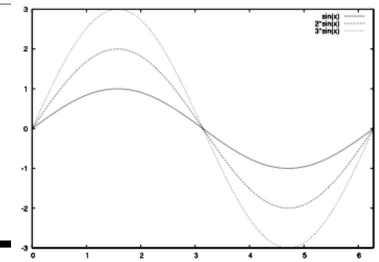
Music and Images

Digital Representation of Analog

The Sine Wave

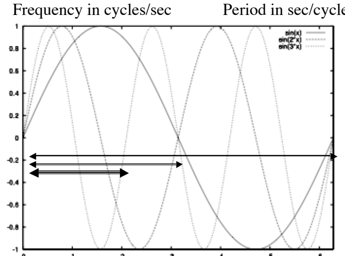


Amplitude



Frequency and Period

Frequency in cycles/sec Period in sec/cycle



Frequency = 1/Period Period = 1/Frequency

Period and Wavelength

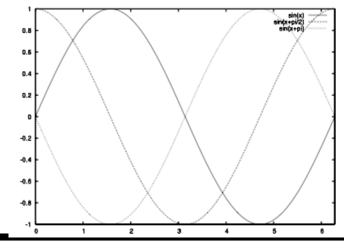
- Period = time duration of one cycle
- Wavelength = spatial length of one cycle
- For waves traveling at a fixed speed, period and wavelength are proportional
- E.g. light travels at speed *c* m/sec, and

  Wavelength = *c* * Period

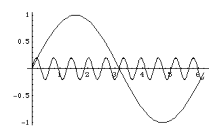
Wavelength · Frequency = Speed
(m) · (#/sec) = m/sec

- If speed is fixed then wavelength and frequency vary inversely
- E.g. speed of light in vacuum, speed of sound in air are constant
- Frequency measured in Hertz: 1 Hz = 1 cycle/sec
- AC current = 60 Hz
- A note above middle C = 440 Hz
- **Audible telephone frequencies = 400 - 3400 Hz = 0.4 - 3.4 KHz**
- Visible light = (4-7.5) · 10^{14} Hz

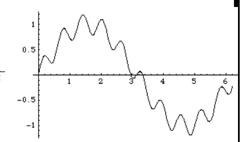
Phase



Sum of Sine Waves

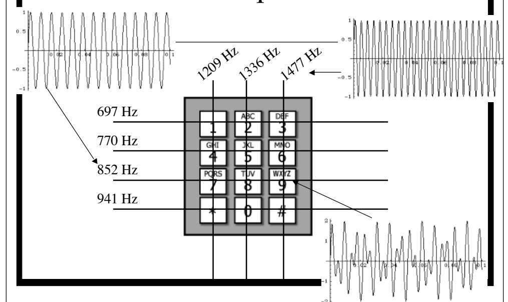


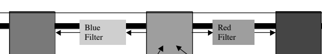
sin(x) and 0.2*sin(10*x)



sin(x) + 0.2*sin(10*x)

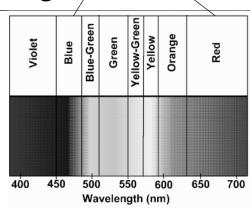
Touch-Tone Telephone





Visible Light

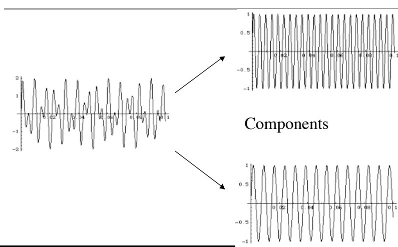
Short wavelength, high frequency



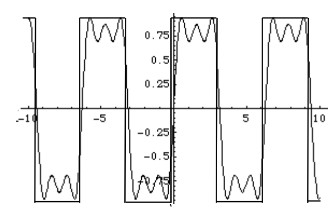
Long wavelength, low frequency

A filter is something that transmits only a limited "band" of wavelengths

Signals can be Filtered

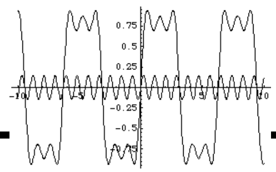


Any Periodic Signal is Approximately a Sum of Sine Waves



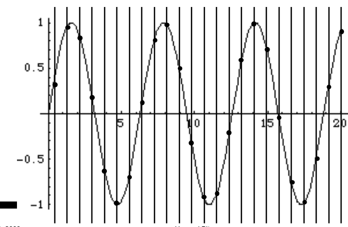
Fourier Analysis = Decomposition of Signal into Sines

- Signal usually is a sum of waves of higher and higher frequency and lower and lower amplitude
- Higher frequency components give greater accuracy
- Next component of square wave:



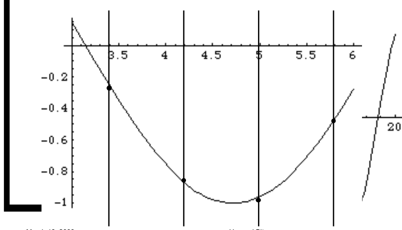
Sampling

A signal can be reconstructed from samples taken at regular intervals as long as the intervals are short enough



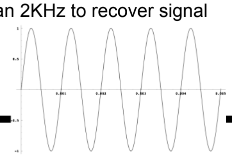
Undersampling causes Aliasing

If the samples are too infrequent a lower-frequency signal may fit the sampled points and the original signal can't be recovered

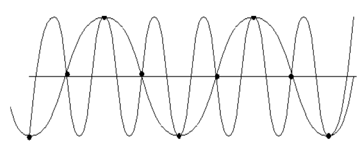


Nyquist Sampling Theorem

- *For the signal to be recovered accurately from the samples, the sampling rate must be more than twice the frequency of the highest-frequency component*
- Wave frequency 1 KHz so sampling must be more than 2KHz to recover signal



Alias = Another Signal with Same Samples as Original

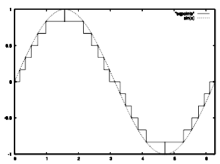


Audio Frequencies and Sampling

- Telephone system designed around 3.4KHz max
- Human hearing up to 20KHz
- Loss of high frequency components ==> poorer quality sound
- Digital telephones sample at 8KHz = 2*4kHz
- CD ROM samples at 44.1KHz > 2*20KHz
- Some PC sound cards sample at this rate
- So VOIP (Voice Over IP) can have higher fidelity than telephone land lines!

Quantization: How Many Bits per Sample?

- *n* bits/sample => 2^n possible sample values



Audio CDs => 16 bits/sample * 2 channels for stereo
Digital Telephones => 8 bits/sample

How Many Bits of Music?

- Audio CD: 1 hour of music =

  3600 s * 44,100 sample/s * 16 bits/sample* 2 stereo channels

  = 5Gb = 636MB
- Bits are used to reconstruct the sine waves, not simply to adjust the volume in jagged jumps

Compression of Music

- CDs are uncompressed
- When CD standard was set it would have been too expensive to put decompression chips into consumer electronics
- Requires intelligence in the processor
- CDs are a dying technology. Already often used only once, to move music onto computer disk or Ipod
- What you can do with information depends on the representation!

Compressing Music Losslessly

- For storage on computer disk, compression is possible because music *samples* have low entropy
- Less space <==> more computing
- Simple example: Take advantage of the fact that successive samples usually differ by only a little
- E.g. Difference coding: Record one value (16 bits) and then just the *changes*, sample to sample
- E.g. 4527; +1, 0, 0, -3, +2, 0, 0, 0, +7, 0, 0, -1, …
- Huffman coding this sequence ==> huge compression
- Real example: FLAC = Free Lossless Audio Code

Lossy Compression of Music

- Once you have the bits, there is lots of computing you can do on them
- Principle: If the average teenager can't hear the difference, why waste money preserving it?
- Rely on psychoacoustic phenomena to compress music in a way that *sounds* almost perfect but isn't
- *Not* to be used at the studio for archival storage
- A family of methods -- depending on the degree of compression, enough information may be thrown away to be subtly audible

Lossy Audio Compession Ideas

- Throw away very high frequency components
- Throw away any component that is soft if it is simultaneous with a loud component
- Change stereo to mono (50% savings) if mostly low frequencies -- where stereo is hard to hear
- MP3, RealAudio, …
- These standards stipulate *decoding* but not *encoding* -- there may be several encodings of the same music that discard different information to produce different storage sizes and bit rates

Still Image and Video Encoding

- GIF and JPEG for still images
- JPEG better for continuous-tone color, GIF for monochrome and line drawings
- JPEG exploits the fact that 24 bits of color are more than the eye can see
- Eye is more sensitive to small fluctuations in intensity than small fluctuations in color
- Spatial coherence: colors similar pixel to pixel
- MPEG exploits temporal coherence for movies: successive frames of video are usually similar