Computer Audio

An Overview
CSE481i
Sound Capstone

(Material freely adapted from sources far too numerous to mention…)

CSE481i Sound Capstone

Capstone course properties:

- work on projects large enough to require teams of several students over a quarter
- apply concepts from more than one area of CSE
- engage in a substantial design effort
- Present work using formal oral presentations and written reports
- produce an interesting, working artifact.

Capstone Schedule

January 12: Teams; project ideas
January 19: Project Proposals
Feb. 16: Alpha versions; team oral reports
March 9: Projects due
March 17: Project demos

Computer Audio

- An interdisciplinary field including
  - Music
  - Computer Science
  - Electrical Engineering (signal processing)
  - Physics (musical acoustics)
  - Psychology (psychoacoustics, music perception)
  - Physiology (hearing, speech and singing)

Computer Audio Areas

- Signal Processing
  - Sound Analysis and Resynthesis
  - Physical Modeling of Musical Instruments and Speech
  - Musical Effects
  - 3D Spatialization
  - Audio Coding and Compression
  - Audio Signal Separation
  - Music Signal Pitch Detection
- AI
  - Machine Recognition of Audio and Music
  - Musical Instrument Recognition
  - Music Perception and Cognition
  - Psychoacoustics
  - AI and Music

Computer Audio Areas

- Software
  - Music Visualization
  - Music Composition Systems and Tools
  - Music Programming Languages
  - Algorithmic Composition
  - Music Notation and Printing
  - Music on the Internet
  - Music in Computer Games
  - Sound Effects in Computer Games
  - Computer Music and Digital Art
- Database
  - Music Information Retrieval
  - Musical Informatics
  - Music Databases

Computer Audio Areas

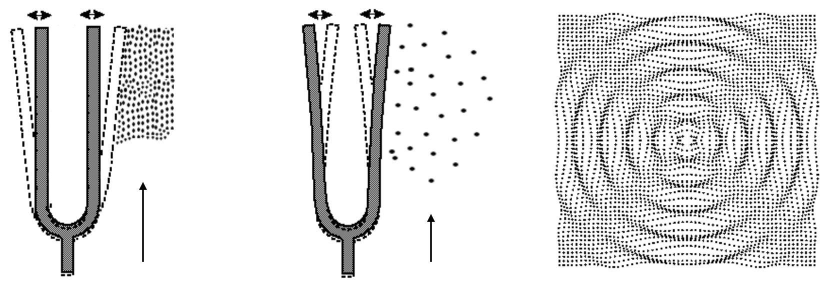
- Computer Engineering
  - Audio Hardware
  - Music Performance Interfaces (new musical instruments)
  - Interactive Performance Systems
  - Real Time Performance Systems
  - Music Workstations
  - Soundcards
  - Music Synthesizers
  - Music and Audio on Mobile Phones
  - Wireless Audio Systems
  - Music Networks
  - MIDI

Computer Audio Areas

- Theory/Science
  - Music Data Structures and Representation
  - Musical Tuning and Intonation
  - Music Grammars
  - Musical Acoustics
  - Acoustics of Musical Instruments and the Voice

What is Sound?

- Variation in air pressure caused by compression and decompression of molecules
- Caused by friction producing force (stick striking symbol, finger plucking guitar string)
- 'Waves' produced by cohesion of molecules, which fall on eardrum or microphone
- Directly and through reflection off surfaces in room
- Ear can detect frequencies in the range 20Hz to 20khZ
- Ear has very high dynamic response compared with eye (ie ability to detect changes in pressure)
- Requires much higher sampling rates to digitize audio compared with images

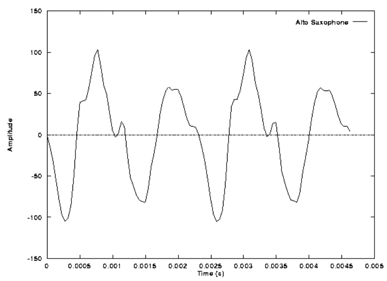


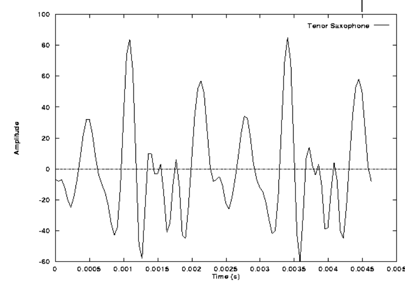
(a) compression (b) rarefaction (c) wave propagation of a tuning fork as seen from above

Properties of sound…

- Waveform – distinctive pattern of variations in air pressure
  - Musical instruments produce orderly repeating waveforms
  - Noise produces random chaotic waveforms
- Fourier demonstrated how any wave form can be decomposed into a series of component sine waves of different frequencies
- Different frequency components, or pure tones, which are added together to produce a complex waveform are called the **frequency spectrum** of that waveform

Same note.. different waveforms





- Both figures show an ‘A’ note, left played on an alto sax and the right on a tenor sax.
- Both have additional frequencies as well as the main 440Hz

Physical and subjective attributes..

- Important to distinguish between the properties of a stimulus and those of a subjective response to that stimulus
- A linear increase in the stimulus value does **not** necessarily produce a similar increase in the subjective response

Stimulus value	Subjective response
(luminance)	(brightness)
Amplitude of wave	Loudness of sound
Frequency of wave	Pitch of sound
Several attributes (hard to define)	Timbre of sound

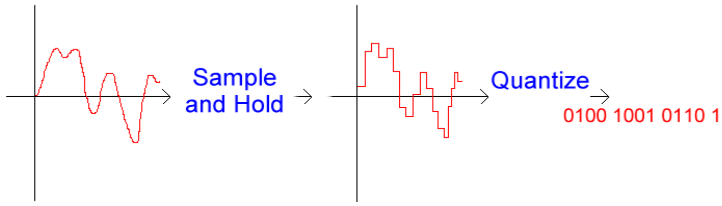
Amplitude and Frequency

- **Amplitude** measured in decibels
  - The louder a sound is, the more it will mask or dominate other other sounds adjacent to it in time
- **Frequency** measured in cycles per second (Hertz – Hz)
  - More digital information required to encode higher frequency sounds, lower pitched sounds degraded less by low sample rates
- **Timbre**
  - loosely defined by 'tone', 'color', 'texture' of sound that enables brain to differentiate one tone from another
  - Affected by acoustic properties of instruments and room

Digitizing sound

- Analog signal is sampled and converted to a series of digital values (A to D converter)
- Digital values later converted back to analog for playback through speakers (D to A conversion)
- Parameters are **frequency** at which samples are taken and the **resolution** of each sample (i.e number of bits used to encode analog signal value
- Nyquist's theorem prescribes minimum sample rate in order to be able to re-construct analog signal
- If maximum frequency in the waveform is n Hz, then minimum sample rate should be 2n Hz

Sampling and Quantizing



- Sampling – process of acquiring an analog signal
- Quantizing – conversion of held signal into sequence of digital values

Sample rates

- If upper range of ear is 20Khz, then there is no need to faithfully reproduce frequency components in signals higher than this.
- **CD quality**: at least 2 x 20KHz = 44.1KHz

- The human voice has few frequency components lower than 100Hz, or higher than 3000Hz - a **bandwidth** of 2900Hz
- **Speech**: at least 2 x 2.9KHz = 8KHz

Sample data rates

- For CD quality,
  - Rate = 44.1Khz (44100 samples per second)
  - Resolution = 16 bits
  - Stereo = 2 channels
- Data rate = 44100 * 16 * 2 bits/second = 1411200 bits/sec
- (10Mb storage for 1 minute of recorded sound)

Examples of data rates and quality

Sample Rate	Resolution	Stereo/Mono	Bytes (1 min)
44.1 KHz	16 bit	Stereo	10.1 Mb
44.1 KHz	8 bit	Mono	2.6 Mb
22.05 KHz	16 bit	Stereo	5.25 Mb
22.05 KHz	8 bit	Mono	1.3 Mb
11 KHz	8 bit	Mono	650 Kb
5.5 KHz	8 bit	Mono	325 Kb

CD quality audio

As good as a TVs audio

As good as a bad phone line

Digitized vs. Synthesized

Multimedia sound comes from two sources:

- Digitized – from an external (sampled) real life sound
- Synthesized – created from waveforms in a sound card for example

Traditional analog sound synthesis is achieved by

- Creating a waveform using an oscillator, which sets the basic frequency
- Adding an "envelope", by specifying parameters such as attack, decay, sustain, release
- Then sending through filter(s) to modify timbre

MIDI – Musical Instruments

- Digital Interface – supported by many instruments/ computers/ manufacturers (1980)
- Defines set of messages indicating note/ instrument/ pitch/ attack etc
- Sound card/ Synthesizer takes this symbolic message and ‘creates’ matching sound
- Sampled sounds can be stored by users on better equipment
- Compare waveforms to bitmapped images, midi to vector graphics

Digital Sound Synthesis Methods

Depending on your age, you might think the first synthesizer looked something like this:

However, it looked more like this:

Synthesis Definition

The Oxford Classical Dictionary defines synthesis as:

- Combination, composition, putting together
- Building up of separate elements into connected whole

Synthesis Definition

- Generally, most people associate synthesis purely with subtractive synthesis
- Very limiting way to look at sound synthesis by electronic means

The Bigger Picture

- Theoretically sound divisions, but practically limiting
- Techniques of different types applicable to others

Subtractive

Sampling

Analog

Granular

Waveshaping

FM

Physical Modeling

Additive

In the beginning…

- Additive synthesis
- Principle first utilized in cathedral organs:

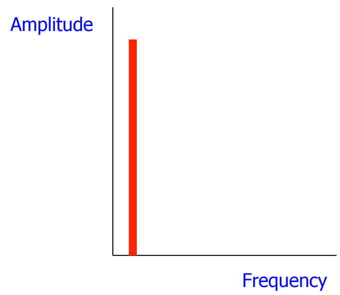
Additive Synthesis

- Mathematical basis:
  - 1822: Jean Baptiste Joseph, Baron de Fourier published theory:
    - *Any arbitrarily complicated periodic waveform can be deconstructed into combinations of sine waves of different amplitudes, frequencies and phases*

  –This is accomplished by the *Fast Fourier Transform*: FFT

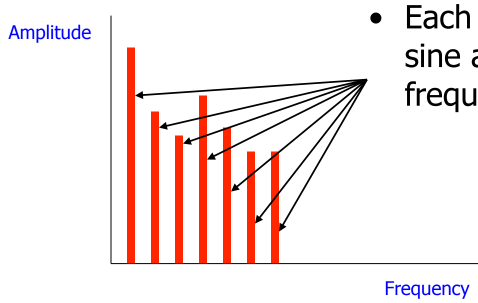
Additive Synthesis

- Sine wave = simplest possible waveform
- Contains only the fundamental



Additive Synthesis

- A more complex waveform will be composed of any number of sines of varying frequencies and amplitudes:

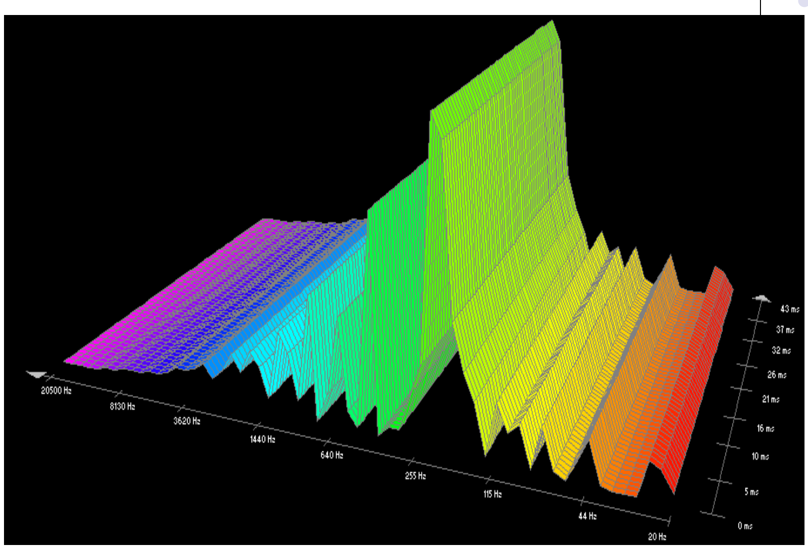


- Each line represents a sine at a specific frequency and amplitude

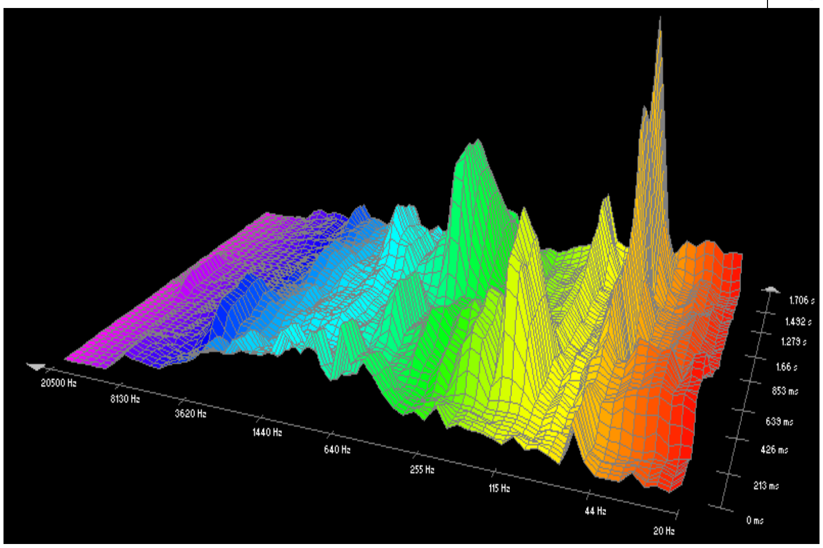
Additive Synthesis

- But this simple approach hides many difficulties
- Theory shown so far deals with a single moment in a sound's duration
- Most sounds are complex and evolving

Sawtooth Wave



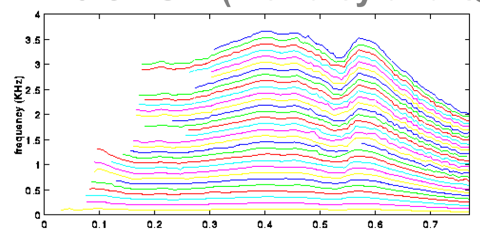
Complex Wave



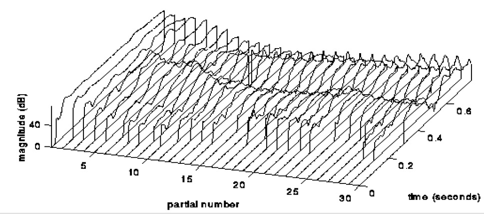
Additive Synthesis

- Thus, will have multiple slices depending on:
  - Length of waveform
  - Rate of change of waveform
- Control data therefore massive
- Very hard to create sounds using additive synthesis
- Holy Grail: Analysis-Based Resynthesis

Sinusoidal Analysis "Tracks" (McAuley and Quatieri)



frequency of partials



magnitude of partials

Sinusoidal Additive Synthesis

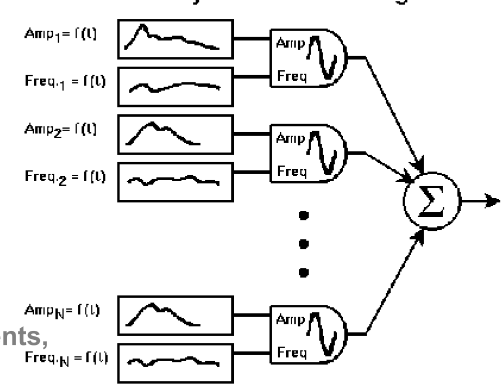
Control the amplitude and frequency of a set of oscillators

The sinusoidal model:

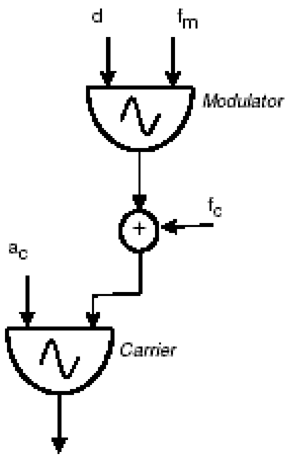
$$s(t) = \sum_{r=1}^{R} A_r(t)\cos[\theta_r(t)]$$

R : number of sinewave components,
$A_r(t)$: instantaneous amplitude,
$\theta_r(t)$: instantaneous phase

Additive Synthesis Block Diagram



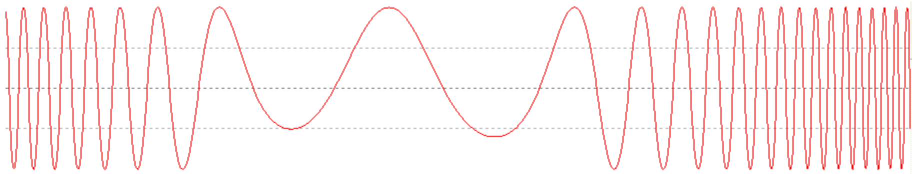
FM Synthesis



- Simple FM: carrier oscillator has its frequency modulated by the output of a modulating oscillator.
- Sidebands produced around carrier at multiples of modulating frequency.
  - Number generated depends on the amplitude of the modulator.

FM Synthesis

- Nothing more than an extreme form of vibrato:



- When the modulation is fast enough, we no longer hear the rise and fall of the vibrato
- Instead, we perceive the changes in pitch as changes in the timbre of the sound

Modulator : Carrier Ratio

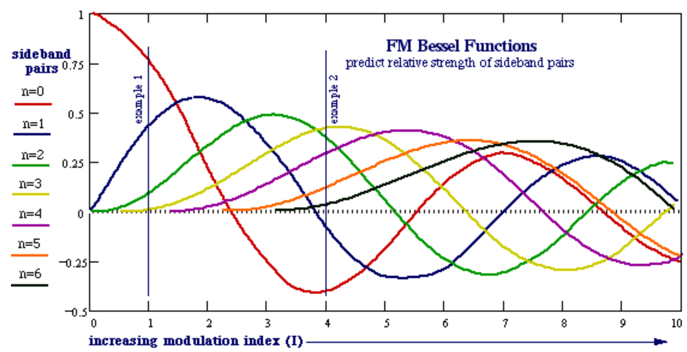
- Sidebands at C + and - (n * Modulator)
- Ratio of M:C determines whether spectrum is harmonic or not.
  - Simple integer ratio = harmonic
  - Non-integer ratio = inharmonic

Modulation Index and Bandwidth

- The *bandwidth* of the FM spectrum is the number of sidebands present.
- The bandwidth is determined by the *Modulation Index*
  - *I = depth of modulation / modulator*
  - *D* depth of modulation, which depends on the amount of amplitude applied to modulating oscillator. ($D = A \times M$)
- If the index is above zero, then sidebands occur.

FM Synthesis

$$J_{(n)}(\beta) = \sum_{k=0}^{\infty} \frac{-1^{k} \cdot \left(\frac{\beta}{2}\right)^{(n+2k)}}{k! \cdot (n+k)!}$$



FM Synthesis

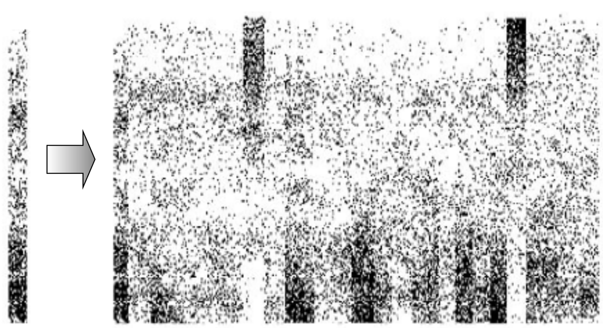
- Unfortunately, the relationship between these is not predictable without experience:
  - as the Index changes, the amplitude of each sideband pair evolves in a different pattern
  - some sidebands gain amplitude, others lose amplitude
  - there may also be cancellation effects caused by phase-inverted sidebands.
- This remains the most significant barrier to learning FM synthesis
- Nevertheless a powerful technique for creating complex sounds

Granular Synthesis

- Attempt to deal with the shortcomings of additive synthesis to deal with changes in the sound over time
- 1947: Dennis Gabor, physicist formulated theory:
  - sound is perceived as a series of short, discrete bursts of energy, each slightly changed in character from the last
- Rooted in quantum physics – coexistence of the wave and photon in light
- Sonic equivalent of the photon is the grain

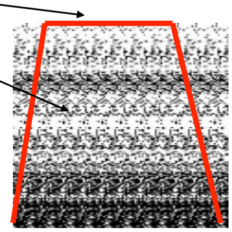
Granular Synthesis

- *Definition*: generation of thousands of short sonic grains which are combined linearly to form large scale audio events
- Grain = tiny piece of sonic data, duration: 10 to 50 ms.



Granular Synthesis

- Two components:
  - Envelope
  - Contents



- NB: Grain Density – number of grains per second
  - Low density leads to rhythmic effects

Granular Synthesis

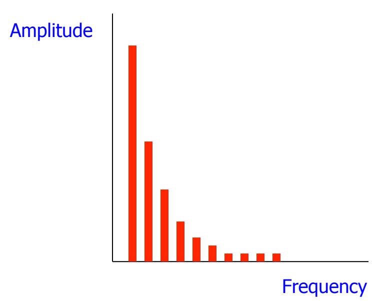
- Subject to same fundamental problem as additive synthesis, though:
  - Tension between precision and control
  - Massive number of grain events
- Basic unit -> grain cloud rather than grain itself
  - Set of rules for generating and controlling grains
- It has some of the drawbacks of FM synthesis as well:
  - Unpredictable results
- But capable of creating sound textures that no other form of synthesis can

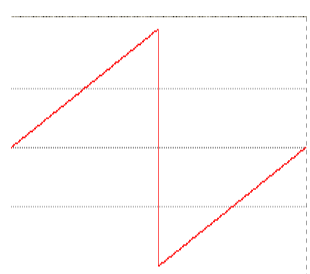
Subtractive Synthesis

- Well understood and widely employed
- Begin with a harmonically rich sound source and remove frequencies by means of filtering
- While any sound source can be employed, traditionally associated with certain waveshapes

Subtractive Synthesis

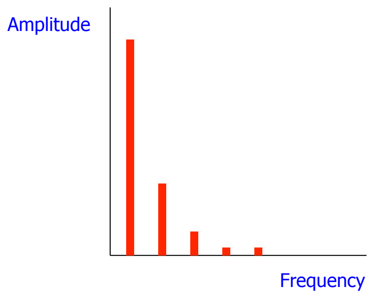
- Sawtooth: contains all harmonics, with amplitude 1/n:

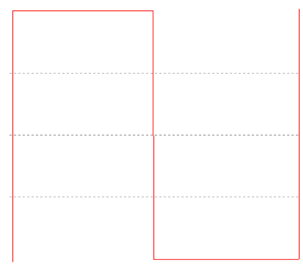




Subtractive Synthesis

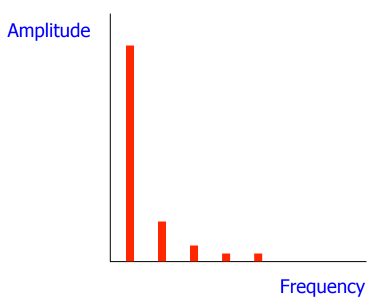
- Square: only odd harmonics present, also with amplitude 1/n

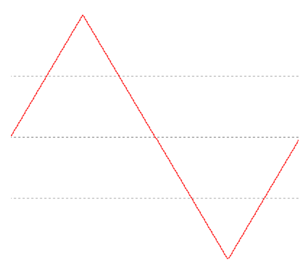




Subtractive Synthesis

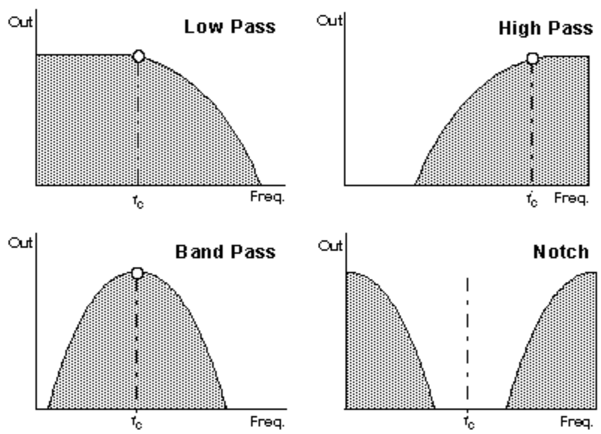
- Triangle: only odd harmonics present, but with amplitude $1/n^2$





Subtractive Synthesis

- Basic Filters



Waveshaping Synthesis

- Sound of a waveform determined primarily by its harmonic content
- Can create new harmonics by passing waveform through non-linear element: *waveshaper*
- Often a Chebyshev polynomial

Physical Modeling

- Modeling sound generation
  - more expressive and realistic sounds
  - ideal for software implementation
  - no need for dedicated hardware
- **Brute force approach**
  - solve equations of motion with respect to boundary conditions
- **Better Approach**
  - Partway solve equations for changing parameters
  - lookup tables
  - lumped processes
  - novel algorithms

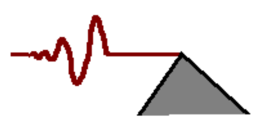
Yamaha VL1

Digital Wave-guide Modeling

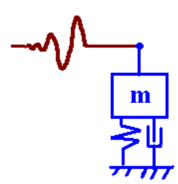
Da-lembert's Solution .. Waves travel in equal an opposite directions

$$f(ct + x) + g(ct - x)$$

Reflected and attenuated at boundary



Boundary behaviour frequency dependant



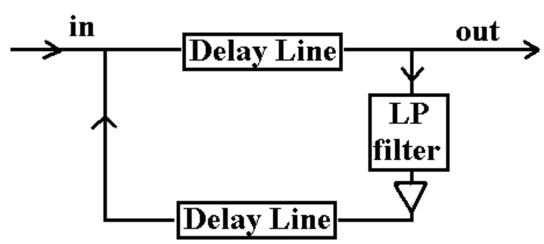
Digital Wave Guide Modeling

Energy in (pluck, hammer, bow, etc)

Delay line simulates time wave travels

Feed back loop simulates reflection

Filter simulates frequency dependant attenuation



Modeling an electric guitar

Array represents wave-guides

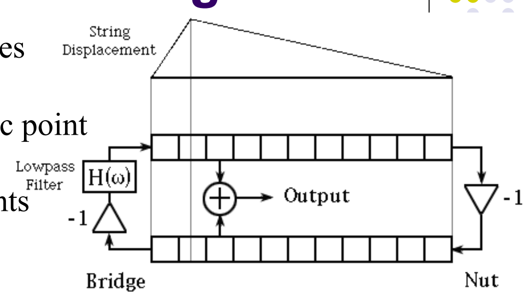
Add displacement at specific point

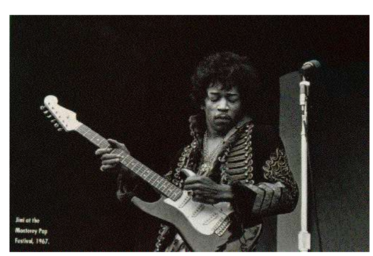
Data passes between elements

Low Pass Filter for bridge

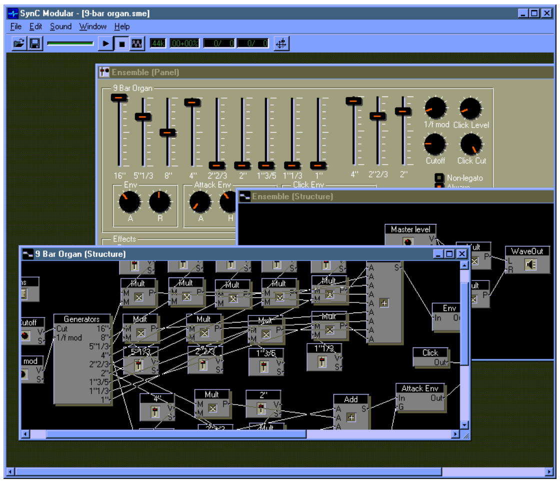
Tap data at pickup point

Can hence delay, modify and add back as feedback





Modular and Virtual Modular Synthesis



SynC modular

A modular paradigm allows for additive, subtractive, fm and sampling synthesis techniques to be used together