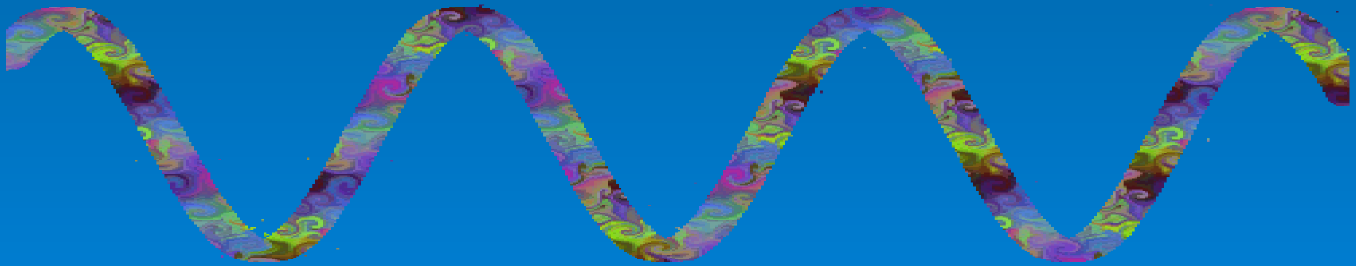
“Audio Synthesis and Sampling”



An introduction to Analog & Digital Audio Generation

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PatternAgents

- Engineering Consultant, Ecospeed, Inc.
  “The world’s best electric assist bicycle system”
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(previous)

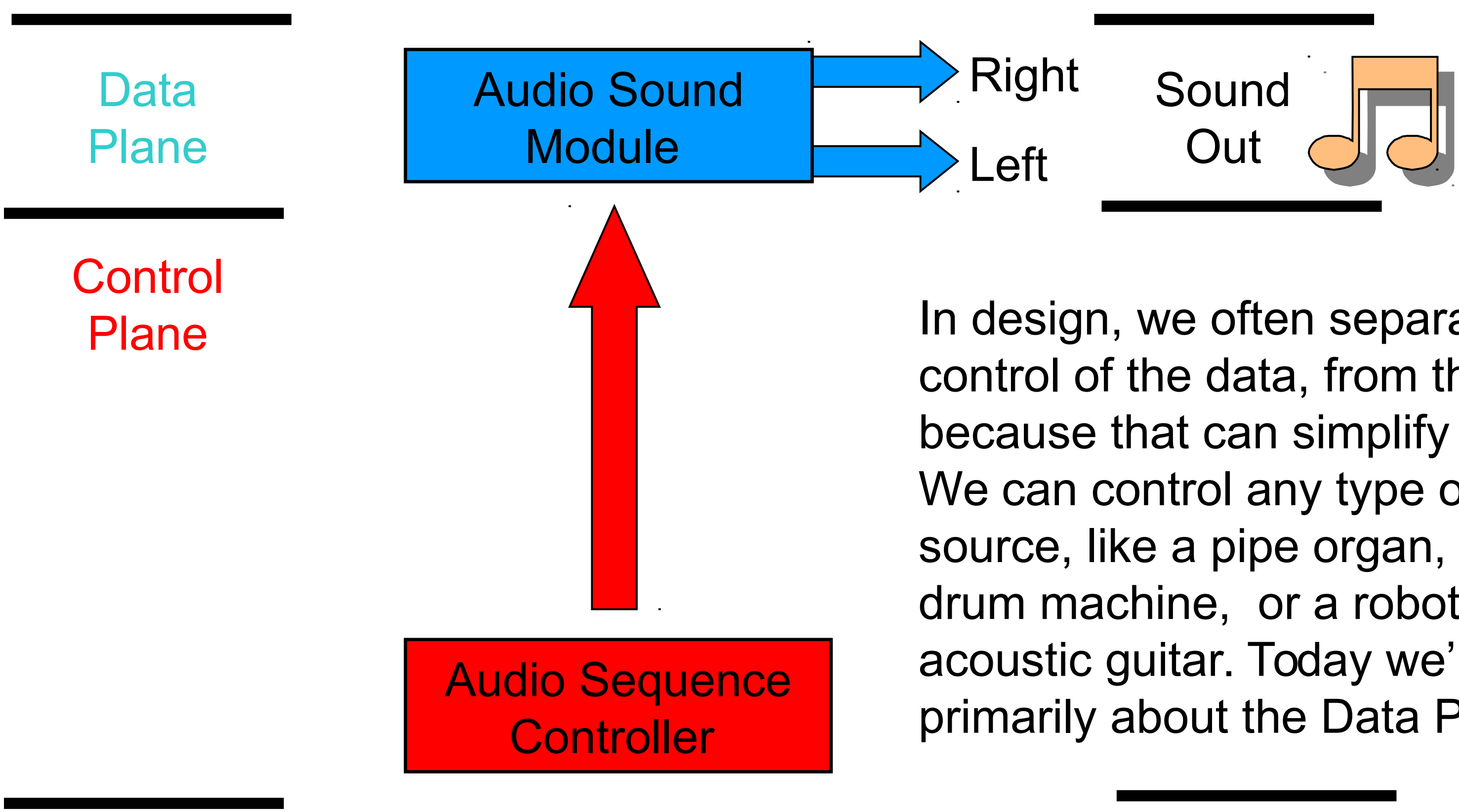
- Design Engineer, EMU Systems, Inc.
  Applied Magic for the Arts
  http://www.creative.com/emu
  (on the design team of one of the first MIDI sampling keyboards)

Source : PatternAgents.com

Agenda

- Basic Electronics Overview
- Subtractive Synthesis Overview
- Other Synthesis Techniques
- Sampling Technology Overview
- Emulator II – MIDI Sampling Keyboard
- Questions & Answers
- Audience Discussion

Control & Data Planes



In design, we often separate the control of the data, from the data itself; because that can simplify the problem. We can control any type of data source, like a pipe organ, synthesizer, drum machine, or a robot playing an acoustic guitar. Today we'll be talking primarily about the Data Plane…

Source : PatternAgents.com

Basic Electronics Overview

- In order to make sure everyone starts on the same page, we'll briefly cover some basic concepts in electronics and physics
- Sometimes it is easier for people to understand the physics of electrons by comparing it to moving water in a hose/pipe as an analogy :

Voltage	- water pressure
Current	- water volume
Resistance	- diameter of the pipe

Source : PatternAgents.com

Ohm’s Law

- Voltage (V) - electric potential (Volts)
- Current (I) - electric charge (Amperes)
- Resistance (R) - material opposition (Ohms)

- Ohm’s Law defines the relationship

$$I = V / R \quad (1 \text{ Amp} = 1 \text{ Volt} / 1 \text{ Ohm})$$

Source : PatternAgents.com

Loudness Measurement

- dB (Decibel) relative logarithmic scale for comparing the loudness of two sources

  dB = log (source1/source2)

- dBm (Decibel milliWatt) logarithmic scale referenced to one milliwatt of power

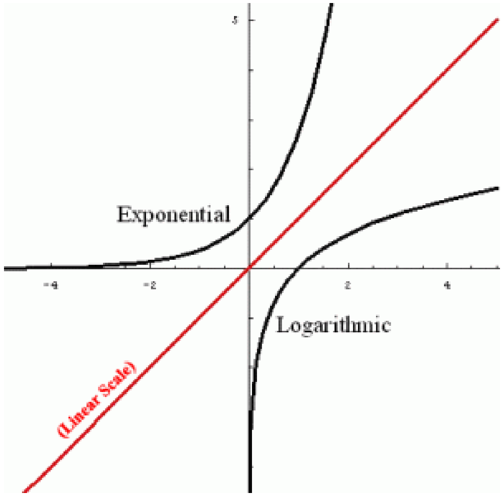
  dBm = 10 * log(source1/1mW)

Source : wikipedia.com

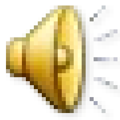
Sound Pressure Levels

- Reference level for air pressure chosen at 20 micropascals (20uPa), about the limit of sensitivity of the human ear

$$SPL = 20 \log (Source1/Source2)$$



What does a one dB difference sound like?

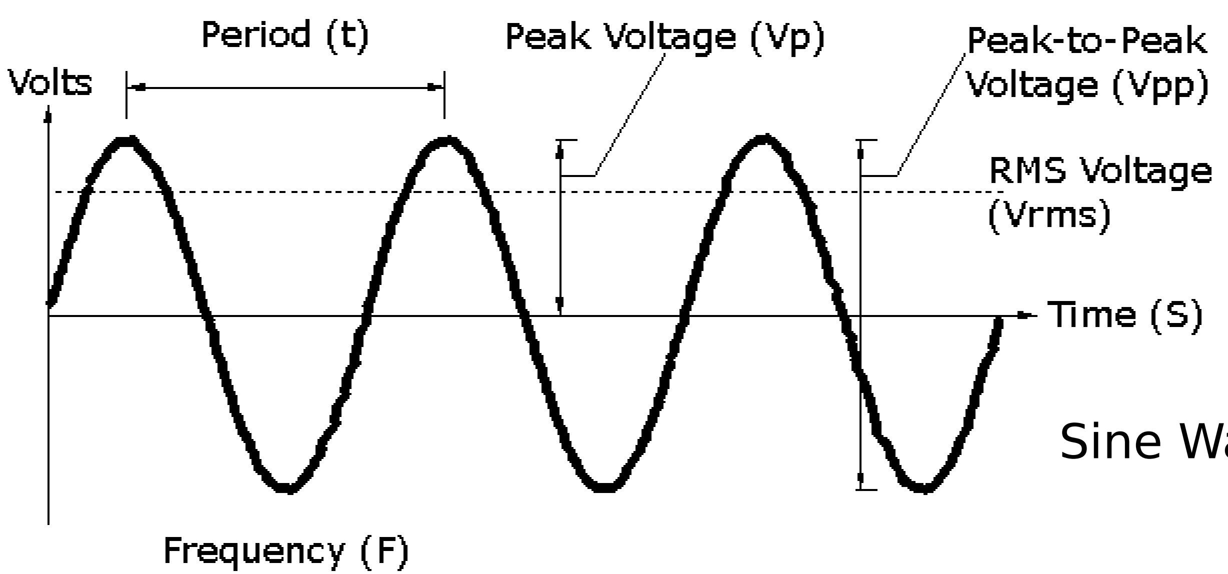


3.0 dB is twice as loud
-3.0 dB is half as loud

Source : UNSW, wikipedia.com

Electronic Waveforms

- Electronic waveforms can be shown graphically in a number of different ways
- Usually they are graphed against Time (S)



Sine Waveform

Source : PatternAgents.com

Waveform Parameters

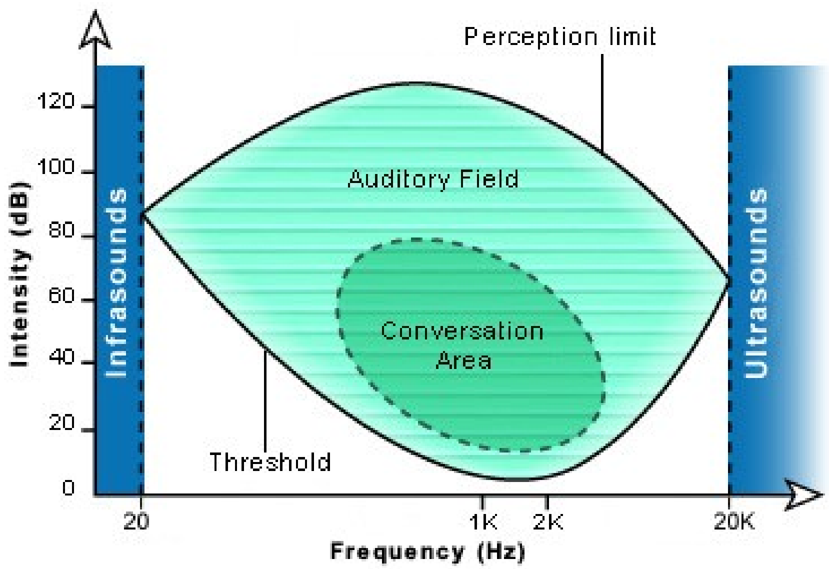
- In a sine wave it is easy to find the primary or fundamental frequency, by measuring the time between the peaks. Period and frequency are merely the inverse of each other

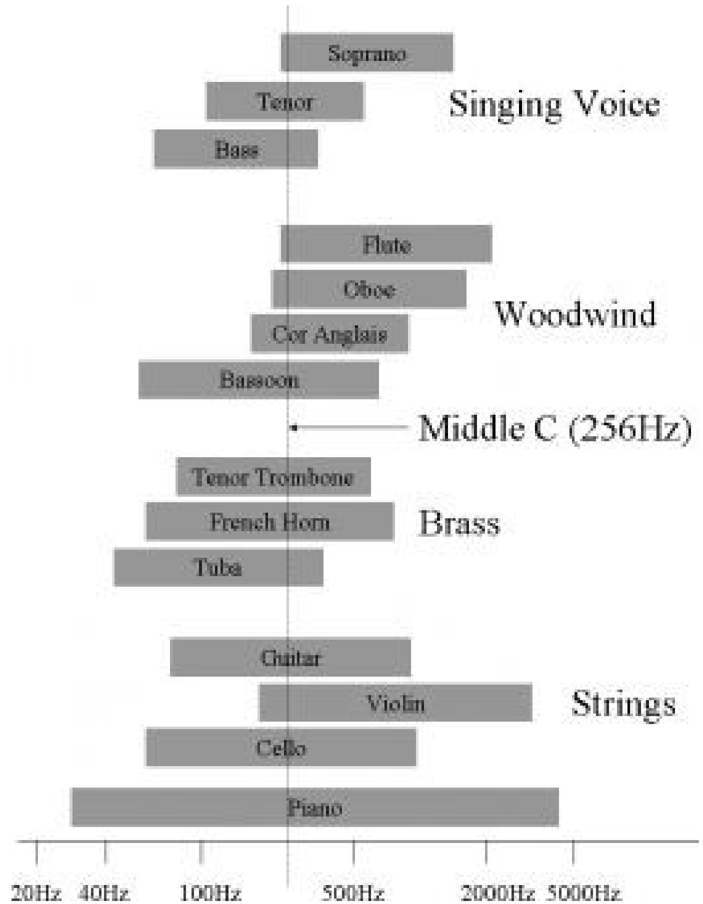
  Frequency (Hertz) = 1 / Period (Sec)

Source : PatternAgents.com

Audio Waveform Frequencies

- Human hearing is generally in the range of 20 hz to 22,000 hz

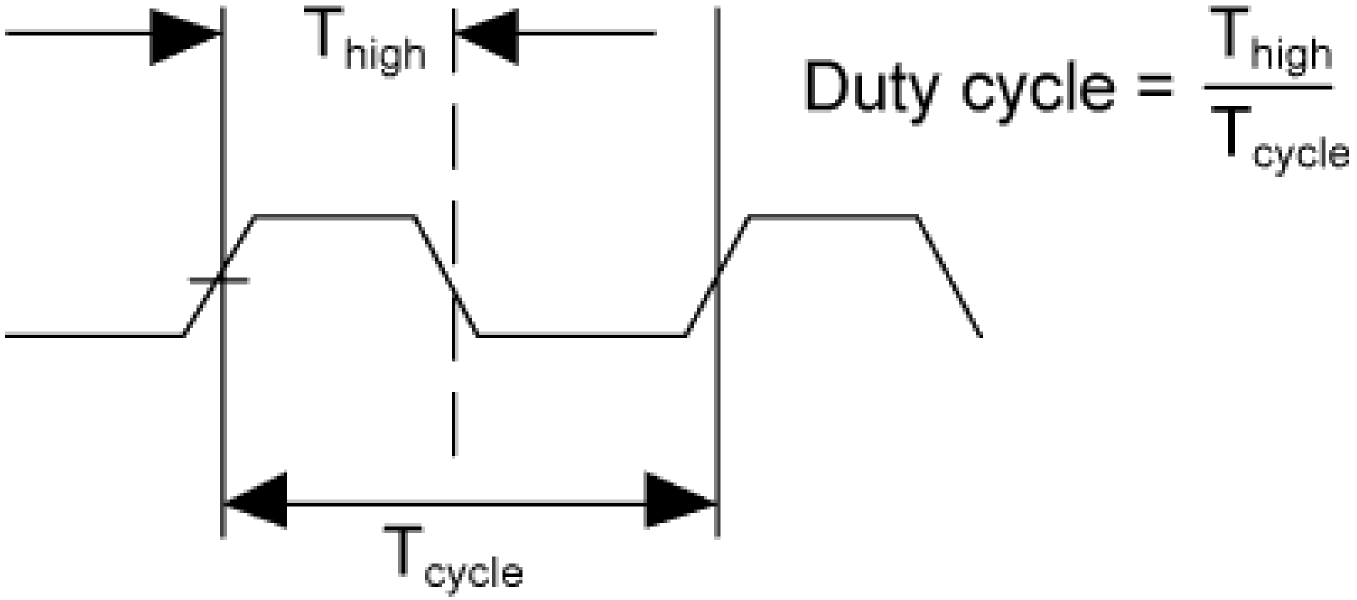




Source : wikipedia.com

Waveform Duty Cycle

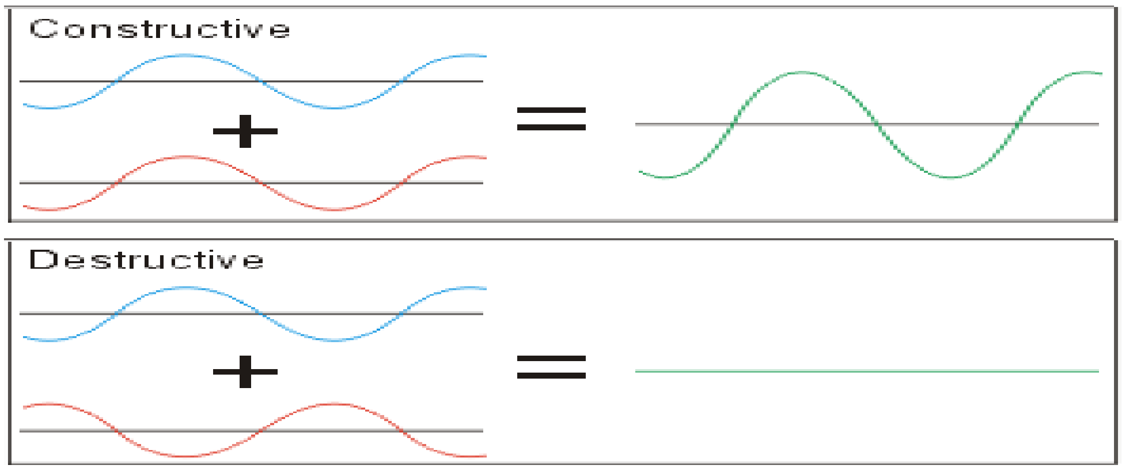
- The Duty Cycle is the percentage of “on” time to the total time of the wave period



Source : wikipedia.com

Waveform Interference

- When multiple waveforms meet (in the air or on a wire…) there can be interference between them, yielding a new waveform

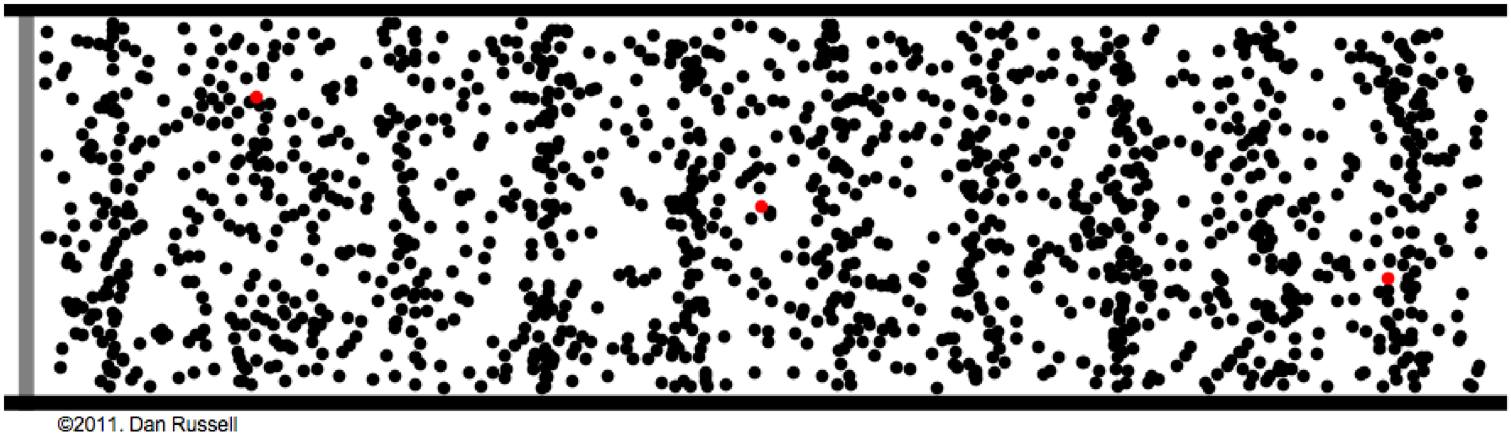


- Destructive Interference is how noise canceling headphones work….

Source : wikipedia.com

Longitudinal Waves

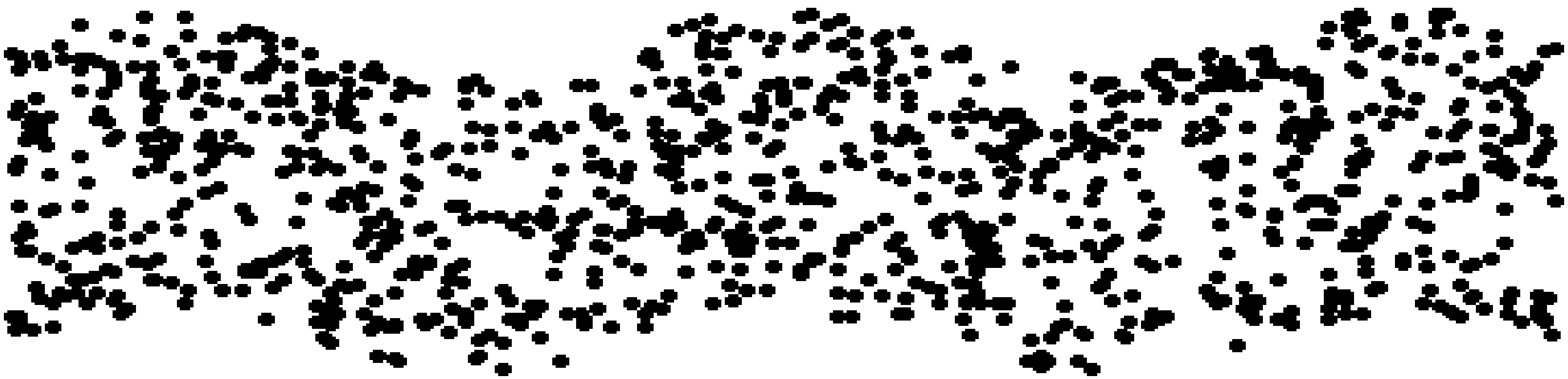
- In a longitudinal wave the energy (or the particle displacement) is parallel to the direction of the wave propagation



Source : wikipedia.com

Transverse Waves

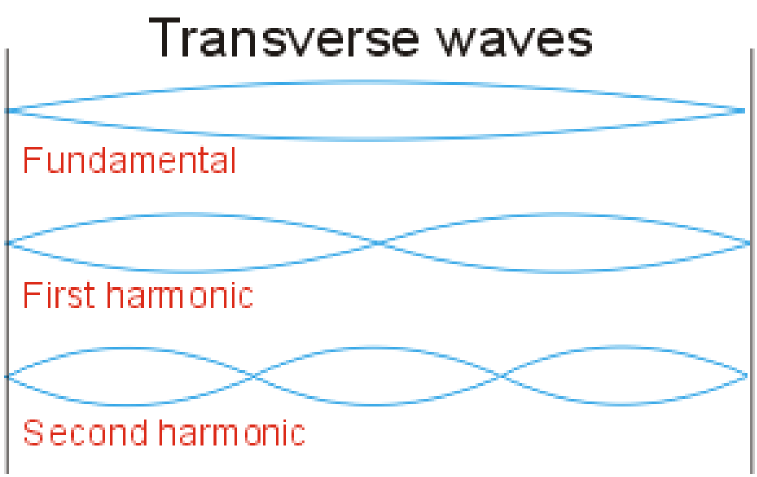
- In a transverse wave the energy (or the particle displacement) is perpendicular to the direction of the wave propagation



Source : wikipedia.com

Harmonics

- Transverse waves can produce harmonics such as when a string that is fixed on both ends is plucked or bowed. Because both ends are fixed, a standing wave pattern with center antinodes can be produced



Source : wikipedia.com

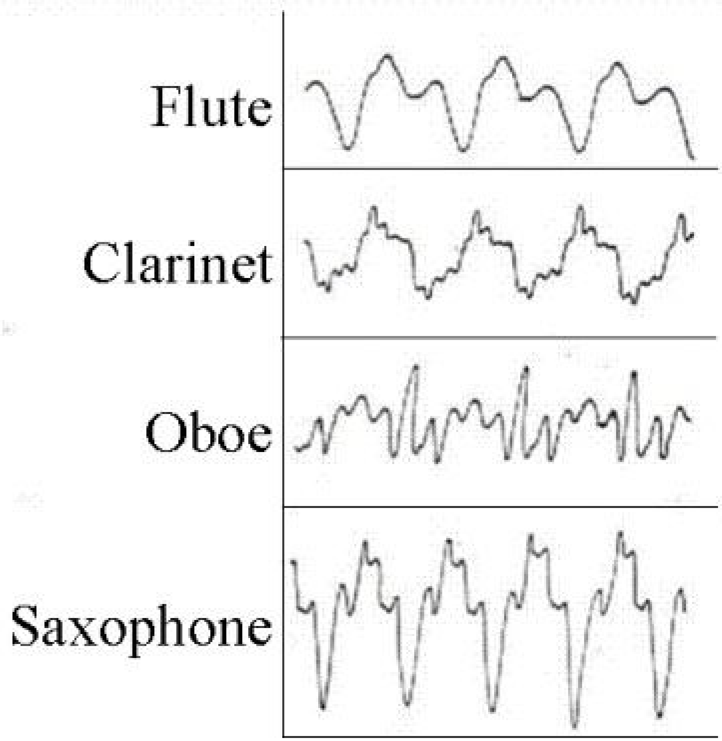
Overtones

- An overtone is any frequency higher than the fundamental frequency of a sound
- Together the fundamental and the overtones are called partials
- Harmonics are partials whose frequencies are integer multiples of the fundamental
- Inharmonics are partials whose frequencies are not integer multiples of the fundamental

Source : wikipedia.com

Complex Waveforms

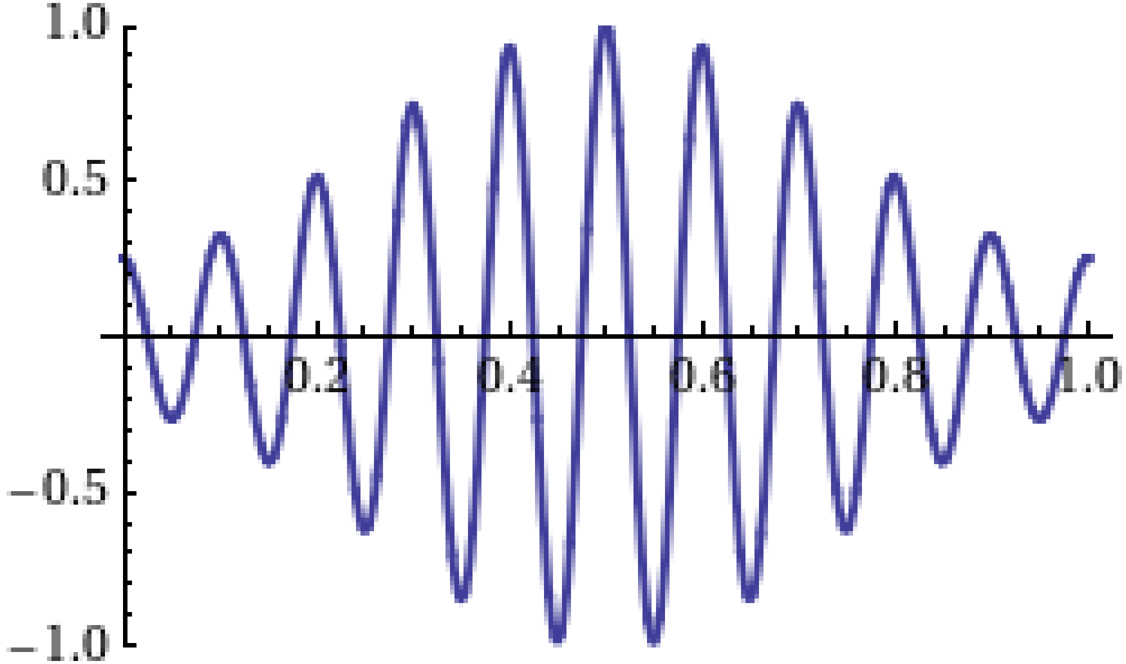
- Most sounds you hear are complex waveforms, they have many overtones



Source : wikipedia.com

Modulating Waveforms

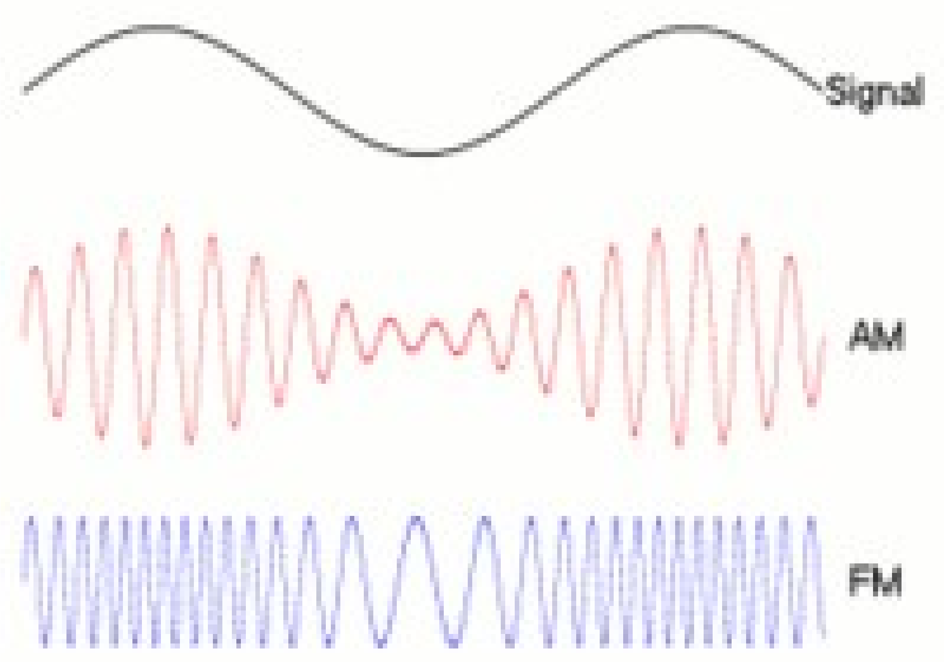
- A basic waveform (carrier wave) can have its parameters modified by another waveform using a process known as modulation



Source : wikipedia.com

Amplitude Modulation

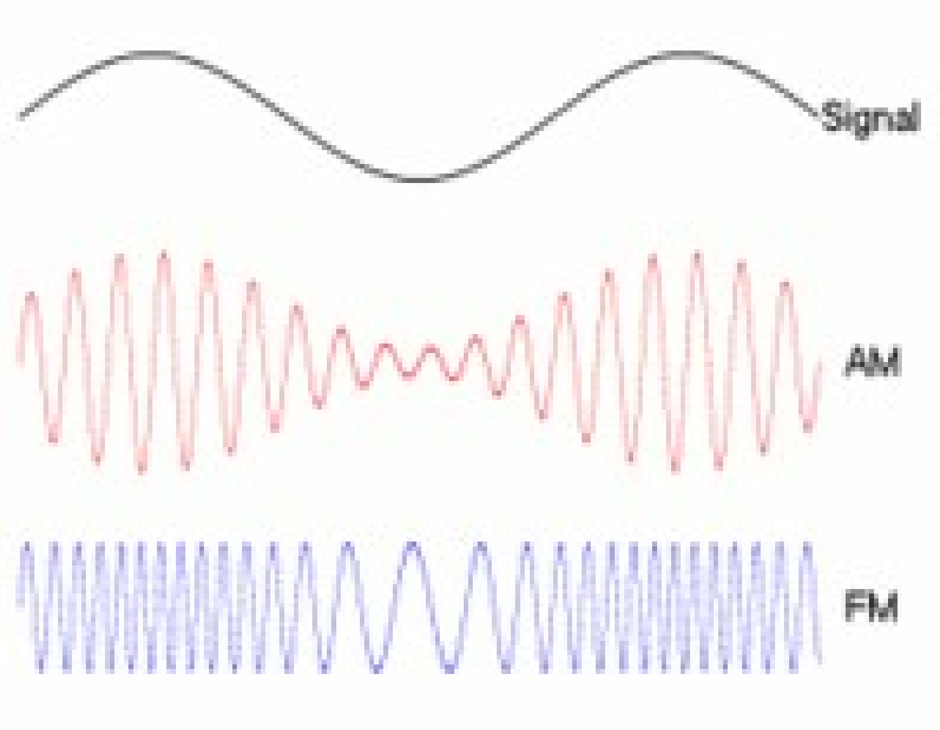
- If the Amplitude of the carrier waveform is changed by another signal it is called Amplitude Modulation



Source : wikipedia.com

Frequency Modulation

- If the Frequency of the carrier waveform is changed by another waveform it is called Frequency Modulation

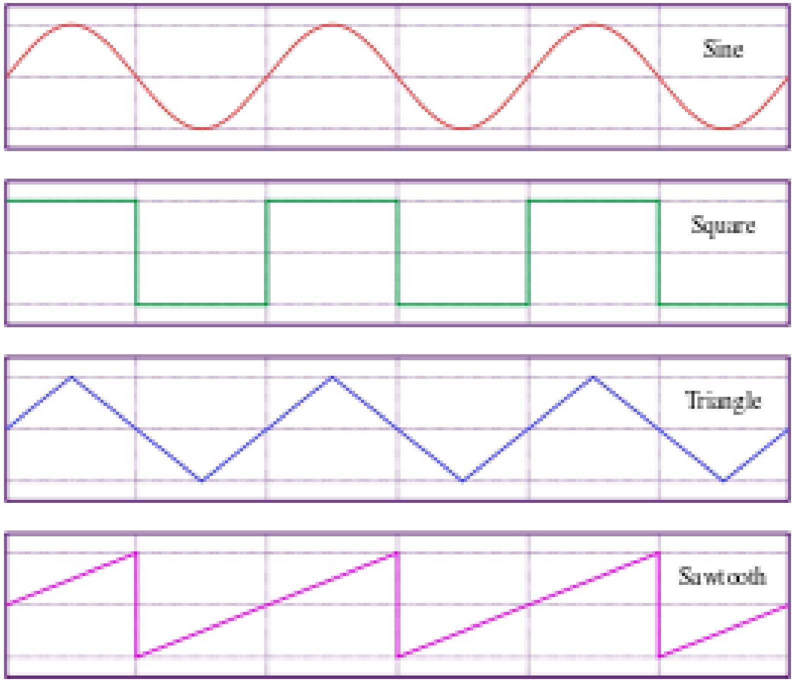


Source : wikipedia.com

Creating Basic Waveforms

- A electrical circuit known as an oscillator is used to create basic waveforms

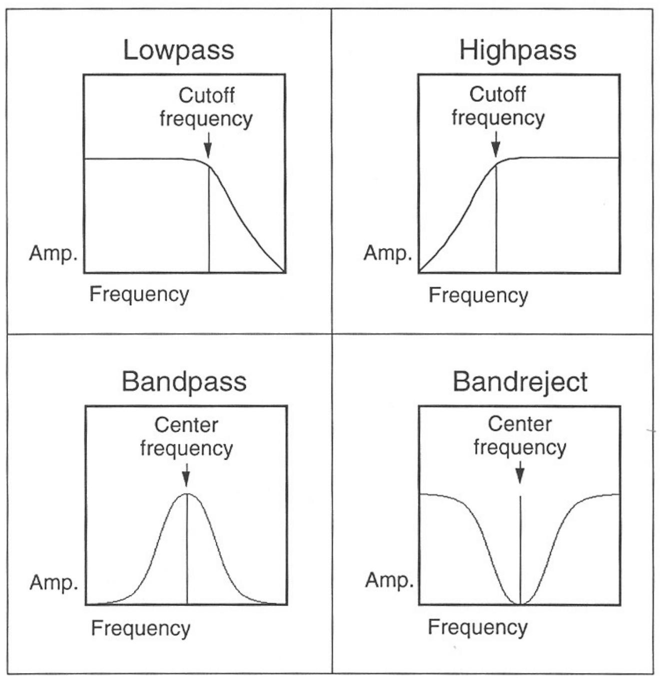
1 Khz 256 Khz



Source : wikipedia.com

Audio Filters

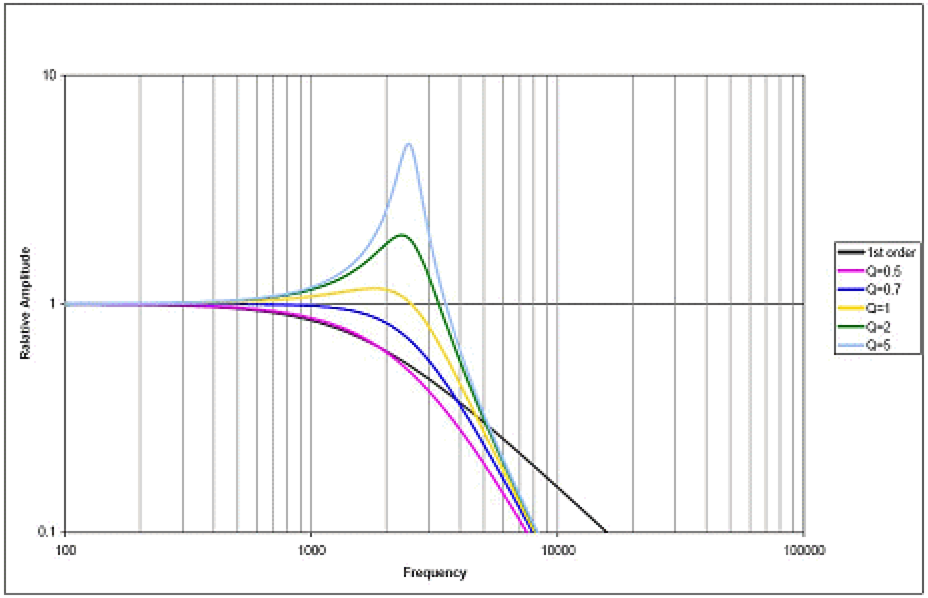
- This is a topic would could spend days on
- An Audio Filter is a frequency dependent amplifier circuit, with several variations



Source : wikipedia.com

Filter Parameters

- Cutoff Frequency : Fc (in Hertz)
- Resonance : Q , Quality Factor (no units)
  a measure of bandwidth relative to frequency
- Several others : Order (2nd/3rd), Type (Elliptical)



Source : Graphic- Beb Hennick/TI

Audio Filter Sounds

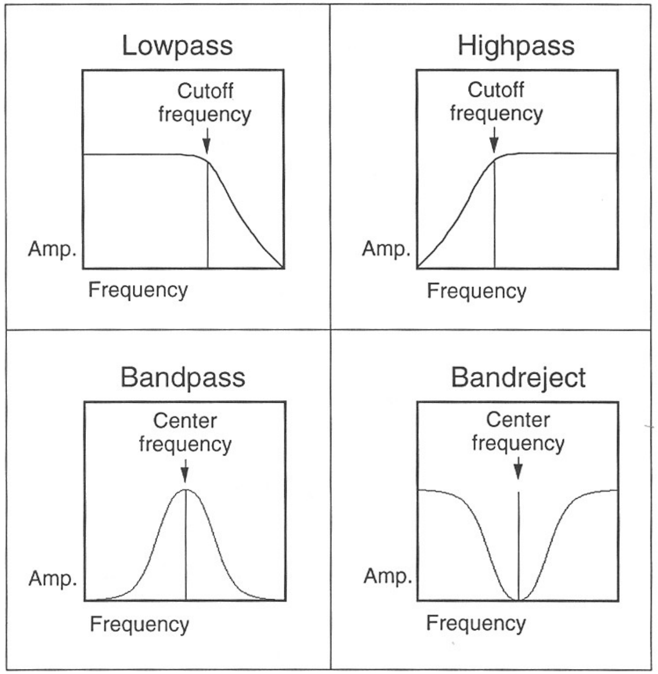
- Okay, so what do they sound like?

LPF – High Q

LPF – Low Q

BPF – High Q

BPF – Low Q



HPF – High Q

HPF – Low Q

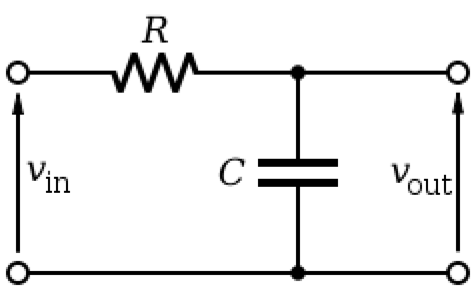
BRF – High Q

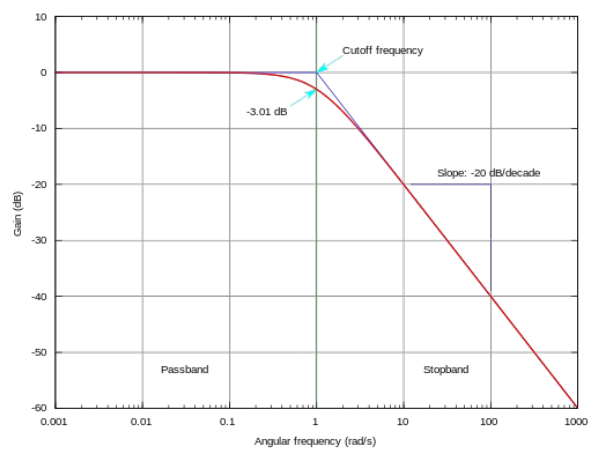
BRF – Low Q

Source : wikipedia.com, Ableton

Low Pass Filter (Passive)

- As the name suggests, only lower frequencies can “pass” thru the filter



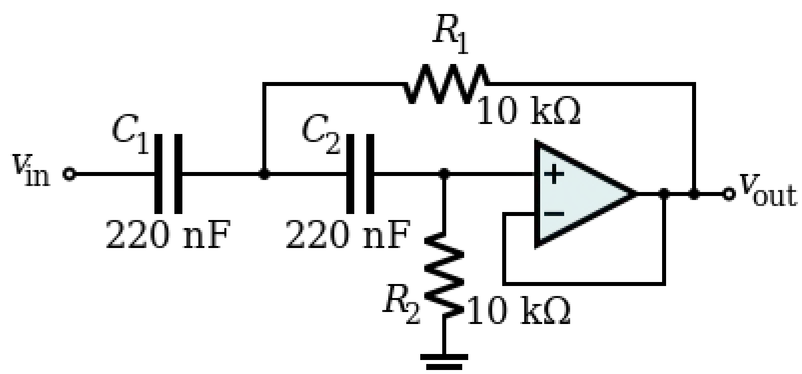


$$f_c = \frac{1}{2\pi R_2 C}$$

Source : wikipedia.com

Sallen-Key Filter (Active)

- Sallen Key is a popular example of an active filter, and uses an operational amplifier (OpAmp) as the active element

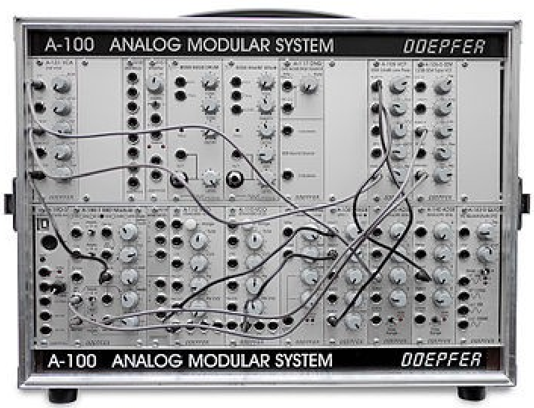


- This LPF has Fc = 15.9Khz and Q = 0.5

Source : wikipedia.com

Voltage Controlled Analog

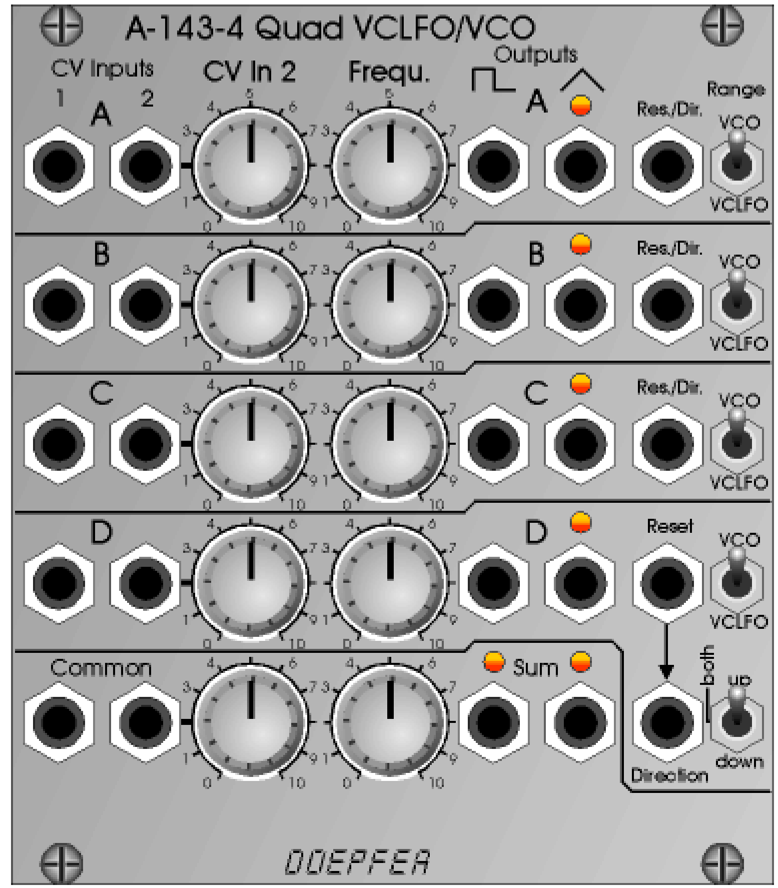
- In the 1960's voltage control of audio functions became popular with the analog synthesizers designed by Dieter Doepfer, Bob Moog, Don Buchla, and many others



Source : wikipedia

Voltage Controlled Oscillators

- In a Analog Voltage Controlled Oscillator (VCO), the frequency of the oscillator output is controlled by an input “Control Voltage”



Source : doepfer.de

Low Frequency Oscillator

- A Low Frequency Oscillator, or LFO is an oscillator that produces a waveform below the human hearing range (< 20Hz).
- LFO's can be used in several ways, either to frequency modulate a carrier signal (Vibrato effect), to amplitude modulate a carrier signal (Tremolo effect) or to control another function, such as a filter center (Wah-Wah effect)

Source : wikipedia.com

LFO Examples

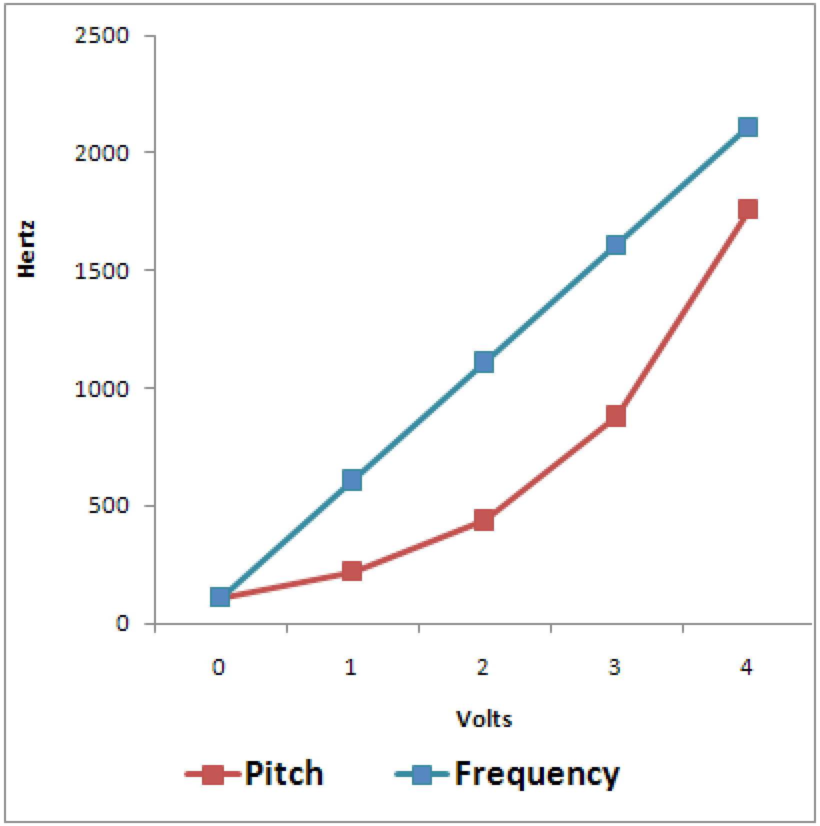
- Tremolo Effect (Amplitude Modulation)
- Wobble Effect (Frequency Modulation)
- Ripple Effect (Filter Modulation)

Source : wikipedia.com

Volt per Octave Frequency

- Voltage Control, or "CV" was often used with Volt per Octave Frequency control.

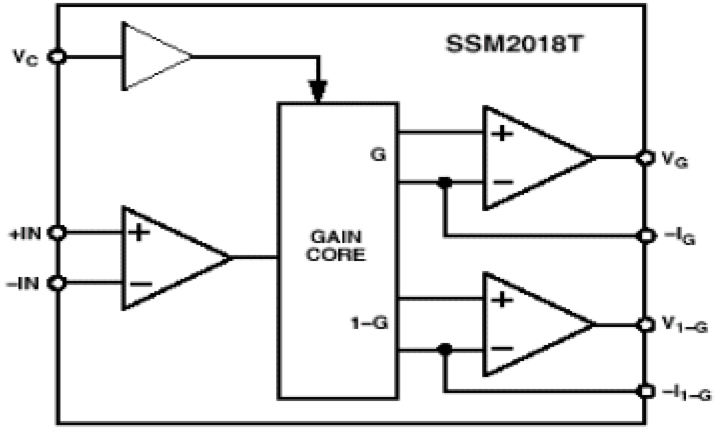
- "Subjective Pitch" can be nonlinear, and is measured in Mels.



Source : emusician.com

Voltage Controlled Amplifier

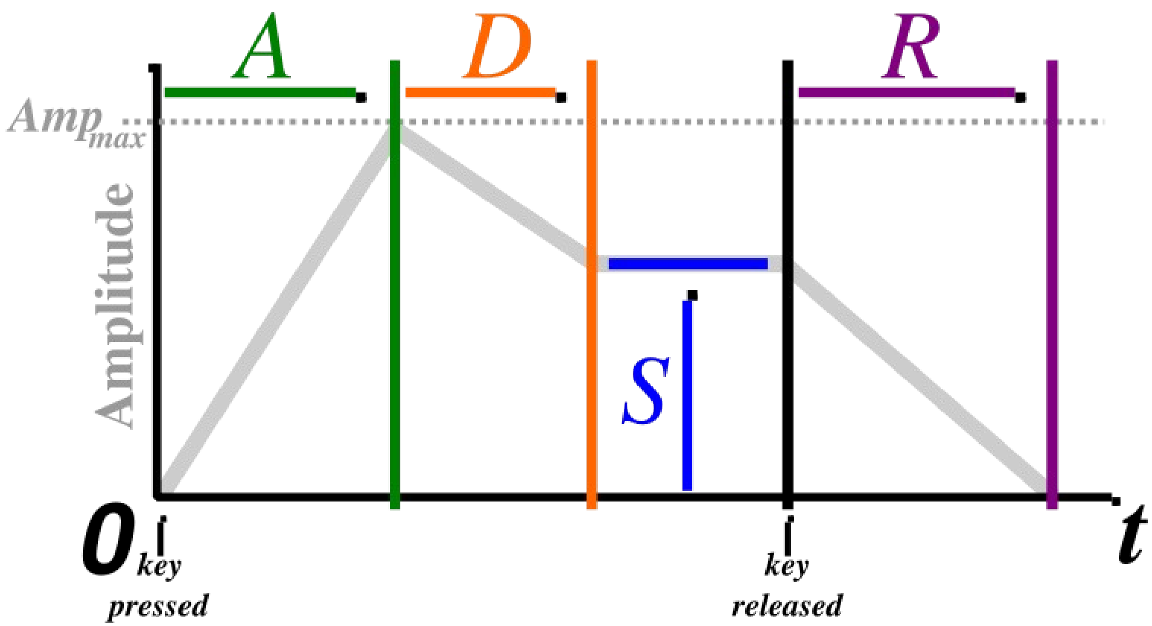
- The Voltage Controlled Amplifier (VCA) is used to vary the volume of the sound based on a control voltage input. The Solid State Music (SSM20xx) parts were widely used for this by Sequential, Oberheim, etc.



Source : analog devices inc.

Waveform Shaping (ADSR)

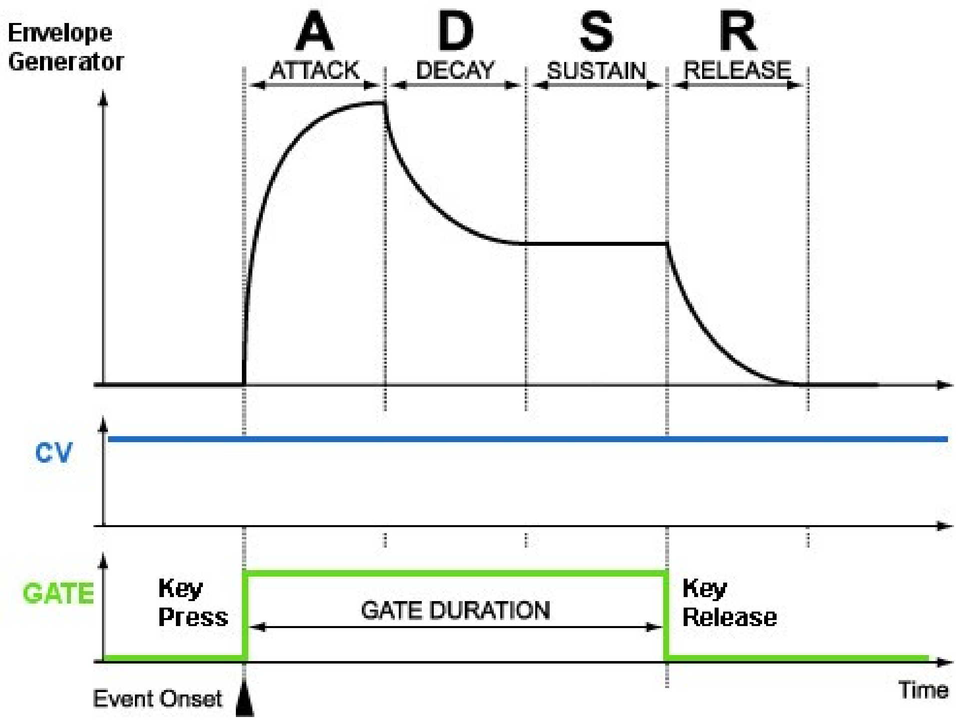
- Several ways to “shape” waveform envelopes to synthesize different sounds, one method is ADSR, which stands for :
  Attack, Decay, Sustain, Release

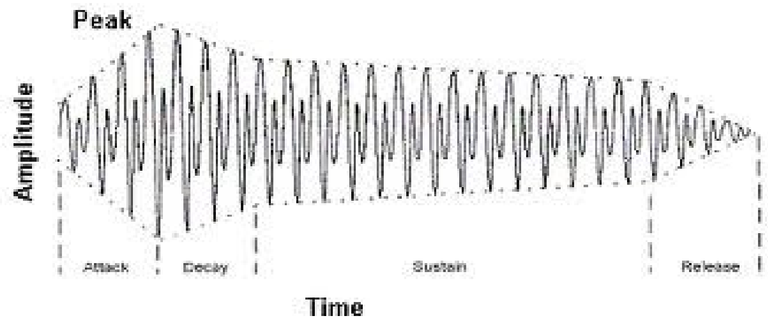


Source : wikipedia.com

ADSR Envelope Generator

- An example of a ADSR Envelope



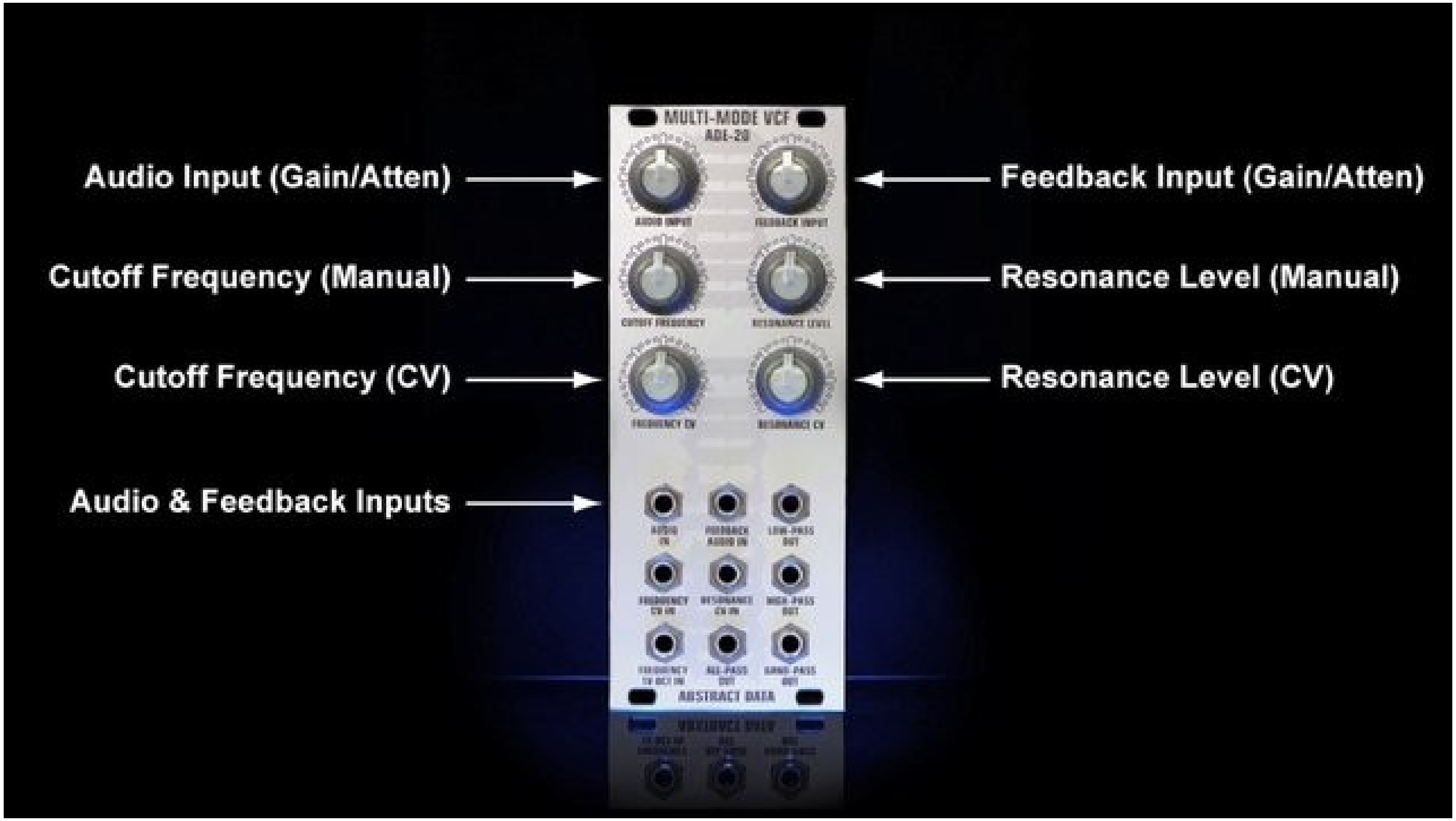


(Keyboard "Gate" Signal)

Source : PatternAgents.com

Voltage Controlled Filter (VCF)

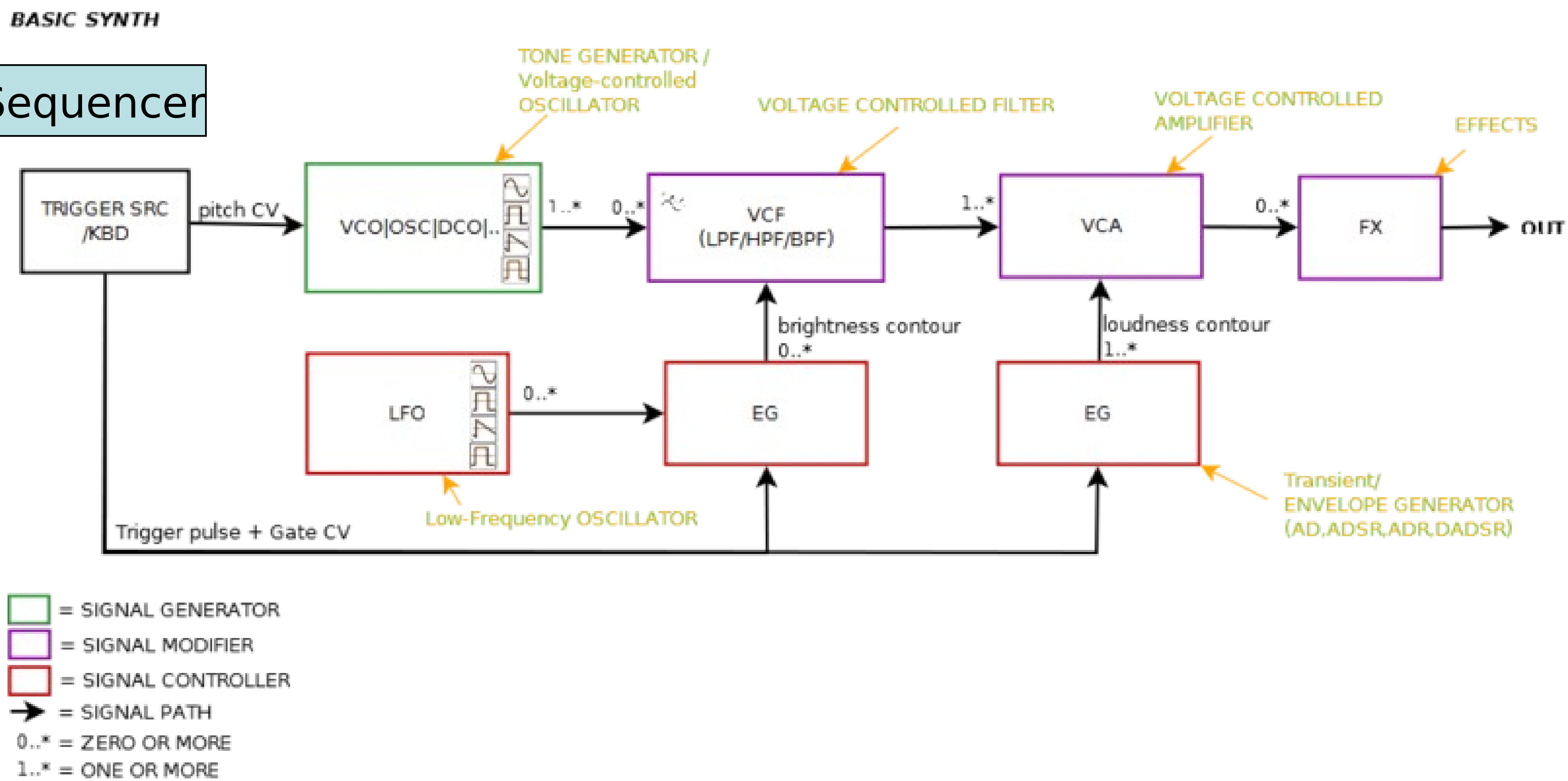
- An example VCF, the ADE-20



Source : Abstract Data Effects

Putting It All Together

Sequencer



Source : wikipedia

CV/Gate Disadvantages

- Wiring : it takes many patch cables
- Compatibility : different voltages
- Calibration : many voltage sources
- Drift : with temperature, tuning issues
- Reliability : too many connections to fail
- Portability : too many connections to rewire
- Patches : each song required rewiring
- Polyphony : difficult for more than a solo voice

Source : PatternAgents.com

SCI Prophet-5

- In the late 1970's "Hybrid" synthesizers started coming available, and they used digital control of patching and routing of CV and gate signals to make a more compact and easily programmable synthesizer

Source : wikipedia

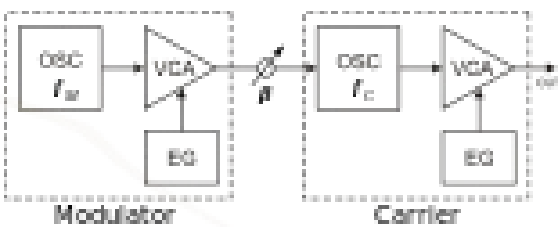
Sequencers

- I won’t have time to cover Sequencers today, as we have covered them in Part 1
- Most people these days are using a program like Garage Band, Ableton, Cubase, or other MIDI program to do their sequencing and patch programming
- Even modular analog synthesizers usually have a MIDI to CV interface for sequences

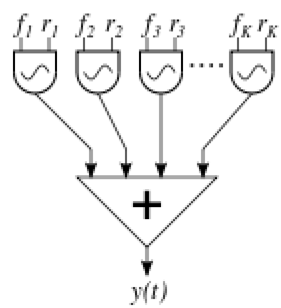
Source : PatternAgents.com

Other Synthesis Techniques

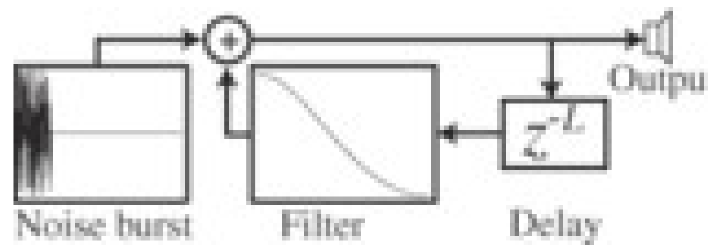
- I also won't have time to cover other technologies today, such as FM synthesis, additive synthesis, or physical modeling



FM Synthesis



Additive Synthesis



Physical Modeling Synthesis

Source : PatternAgents.com

Pure Digital/Computer Based

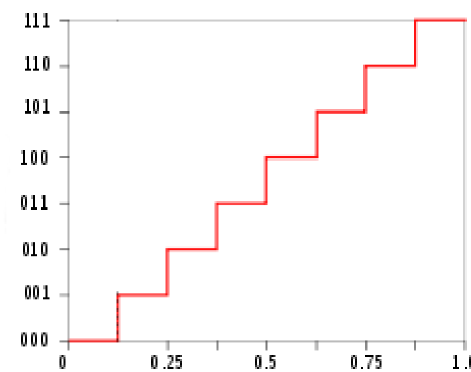
- Early computer music programs stated in the 1950's; and by the 1970's many universities and groups were using mini-computers for direct sound generation, the beginning of widely available Digital Signal Processing (a topic in itself…)

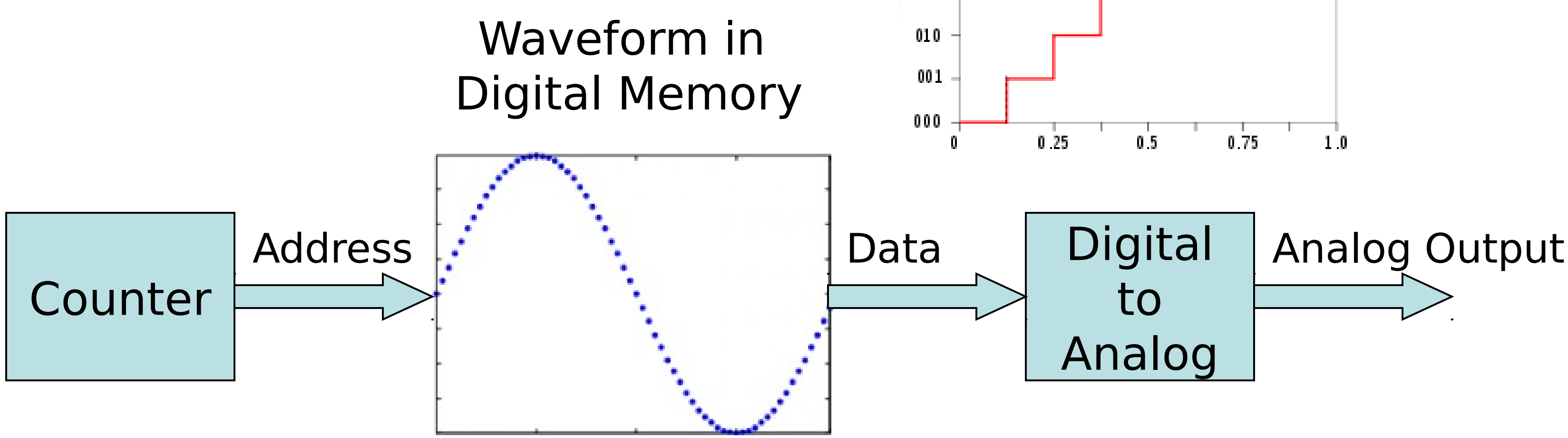
DEC PDP/11 Computer

Source : wikipedia

Digital Waveform Generators

- Digital waveform generators use a stored table of values that when read out to a DAC, then create the desired waveform





Source : wikipedia

Romplers

- Some of the first digital keyboards used prerecorded sounds, often stored on ROM cartridges like early video games. These keyboards and drum machines were known as “ROMplers”. This was because RAM memory was still more expensive than ROM in the 1970’s and 1980’s, and a lower cost instrument could be made.

Source : wikipedia

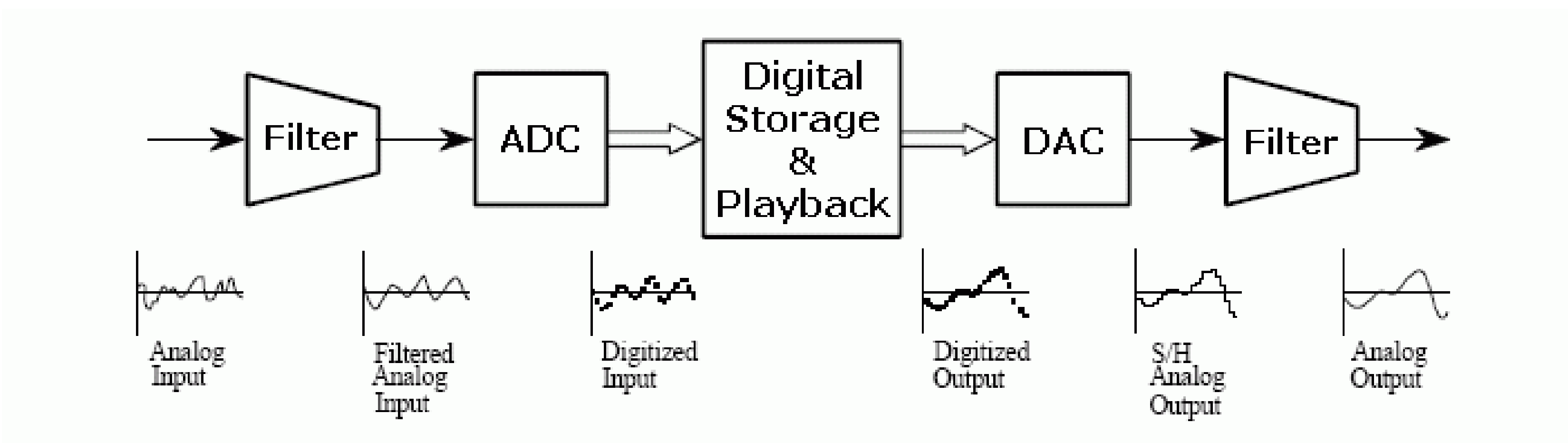
Linn Drum Machine

- Roger Linn introduced the LM-1 drum machine in 1980 at a list price of $5,995. and it was an instant success (Michael Jackson, Human League, Peter Gabriel, etc.)
- The LM-1 was a “rompler” and was the first drum machine to use digital recordings of real acoustic drums

Source : wikipedia

Digital Samplers

- Where most analog synthesizers create waveforms using subtractive synthesis, Digital Samplers record an analog input waveform, and then can play it back



Source : wikipedia

E-mu Emulator I

- E-mu had been a modular analog synthesizer maker thru the 1970's
- E-mu was one of the first to deliver an affordable digital sampling keyboard in 1980, at a price of $7,995 where the early samplers of the time cost $40,000-$100K+

Source : E-mu Systems, Inc.

E-mu Drumulator

- The E-mu Drumulator, a $995 “rompler” drum machine was introduced in early 1983 to wide success that fueled the expansion of the company

Source : E-mu Systems, Inc.

E-mu Emulator II

- We started design work on the “Emulator II” in the fall of 1983. It was a diverse group of designers, and we were aiming to create the very best sampling keyboard we could in the $8K price range (which was very tough to do in the early 1980’s)

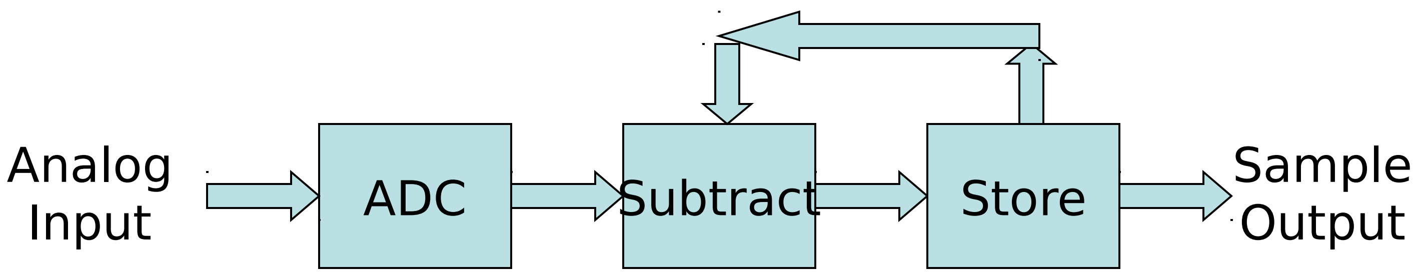
Enjoy the Silence

Sledgehammer

Source : E-mu Systems, Inc.

ADPCM Encoding

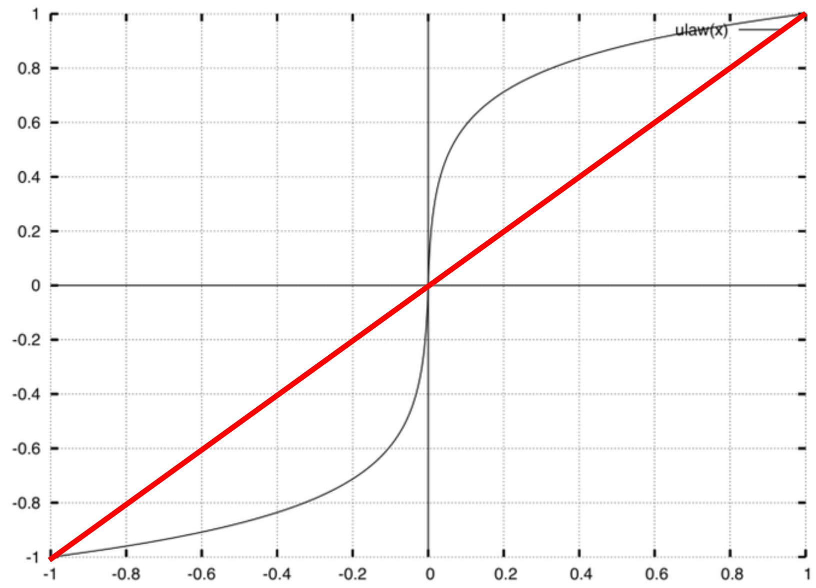
- Since 16-bit Analog-to-Digital Converters were still very expensive in 1983, we used a trick known as Adaptive Differential Pulse Code Modulation, where only the difference between sample values are stored, which makes them easier to fit into 8-bits



Source : E-mu Systems, Inc.

Companding ADC/DAC

- Most ADC & DAC components are linear coded (Red), however we chose to use ulaw coded (Black) ADC & DAC for the EII



Source : E-mu Systems, Inc.

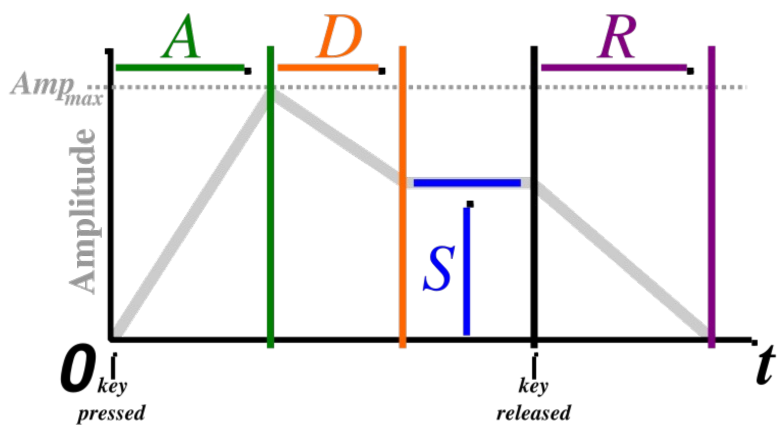
ulaw Compression

- In audio, low frequency signals are more common, so that the combination of ulaw coding and differential quantization meant that the numbers produced remained small.
- This allowed us to use only 8 bit samples to keep the cost of sample storage memory low
- However, the results were comparable with 14 to 16 bit linear encoding, so that the system still sounds very good even today

Source : E-mu Systems, Inc.

“Envelope Sampling”

- Remember how ADSR envelope generation work in analog synthesizers ?

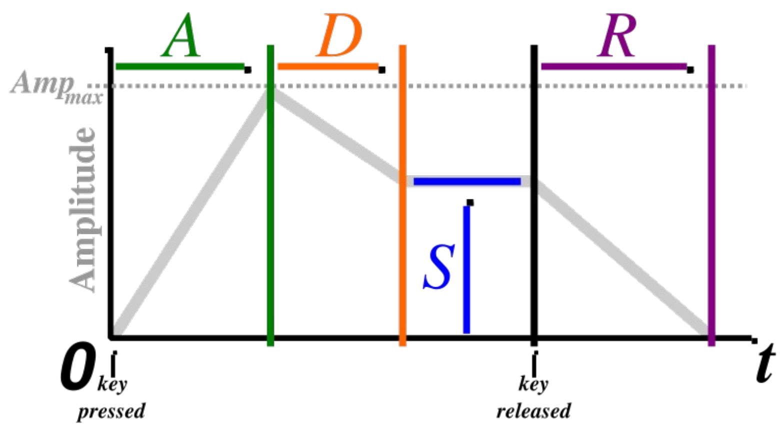


- What if we applied the same concept to digital sampling?

Source : E-mu Systems, Inc.

Envelope Wavetables

- What we did was to create a data structure that could index different samples for ADSR



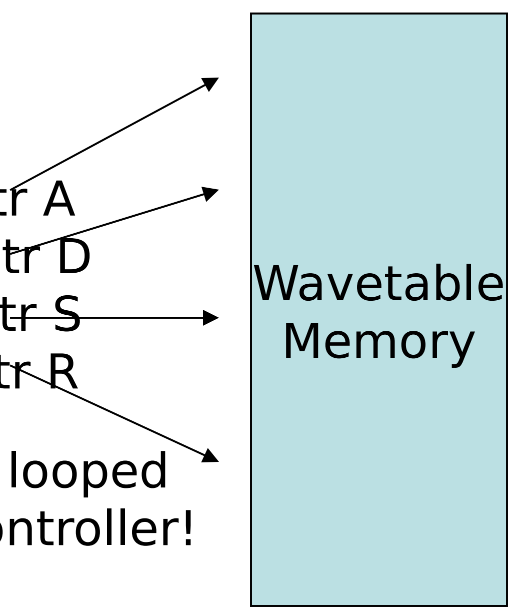
- Now a musician could create a new sound with the Attack of a Drum, Decay of an Oboe, Sustain of a Piano, and the Decay of a Flute

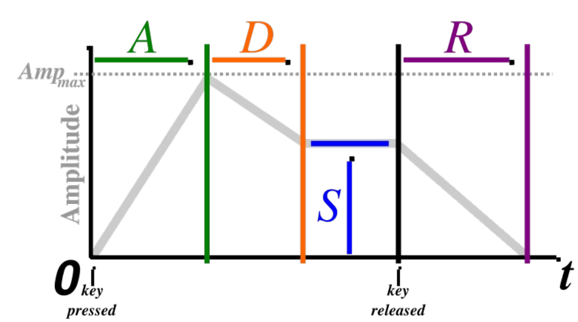
Source : E-mu Systems, Inc.

Emulator II Wavetable Microcode

- This was a unique and novel approach that had never been tried before, and saved enormous amounts of memory, allowing us to reduce the cost of the EII even more

Attack - Memory Ptr A
Decay - Memory Ptr D
Sustain - Memory Ptr S
Release - Memory Ptr R

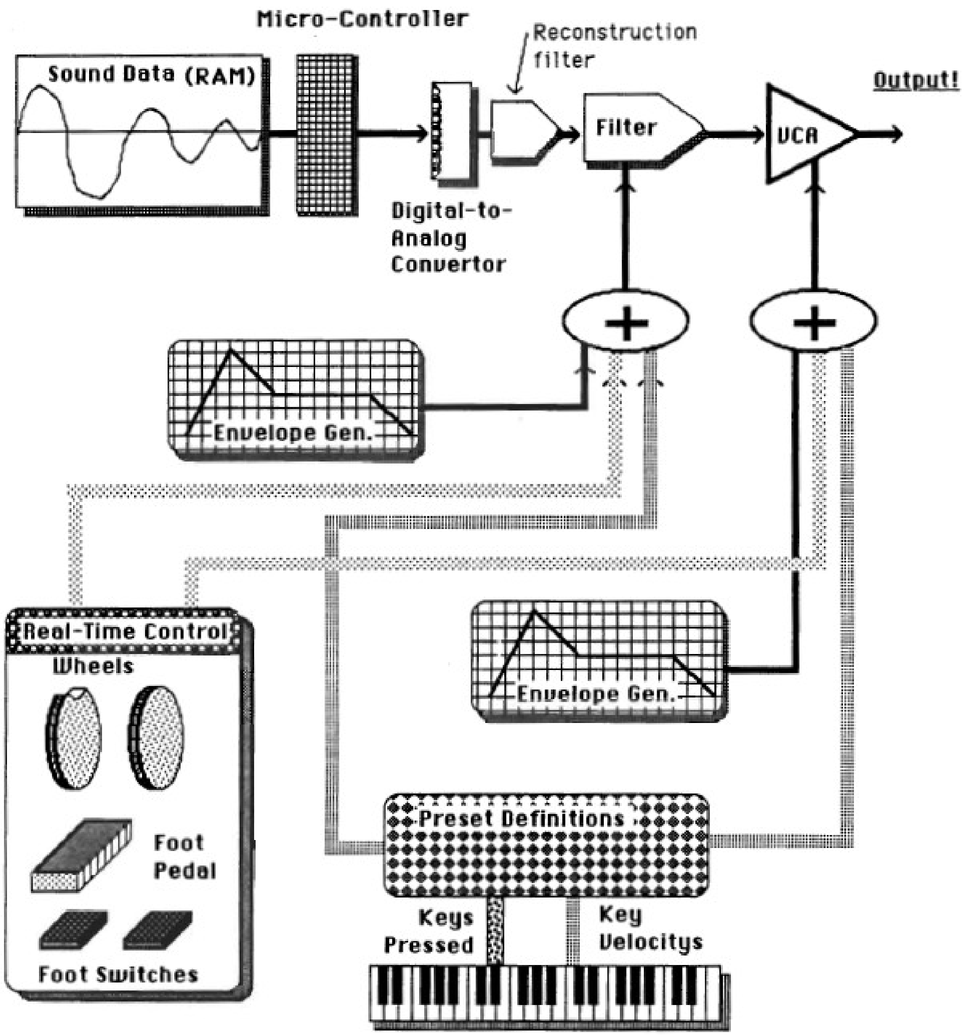




Any of these could be looped using the hardware controller!

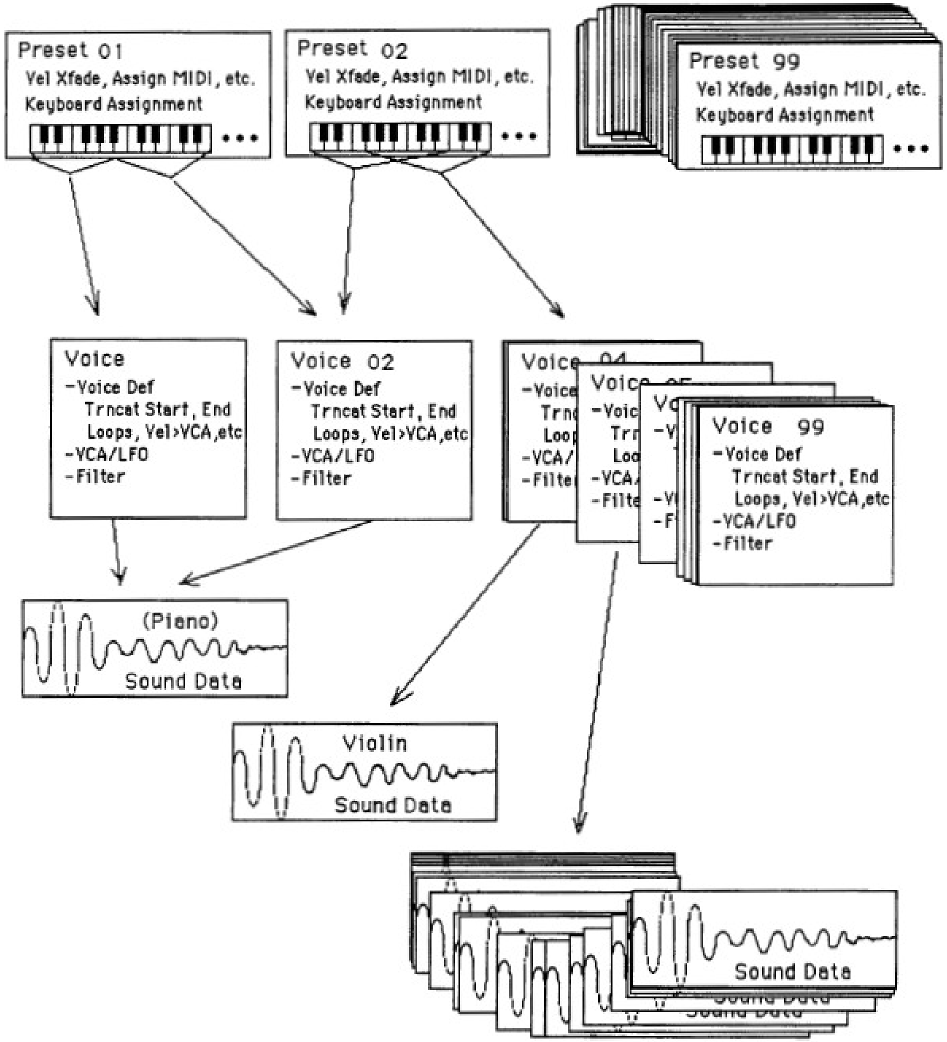
Source : E-mu Systems, Inc.

Emulator II Voice



Source : Dana Massie, E-mu Systems, Inc.

Emulator II Sample Banks



Source : Dana Massie, E-mu Systems, Inc.

Emulator II Real Time Controls

- Another innovation was in the patching of Real Time Controls, like pitch wheel, volume pedal, sustain pedal, etc.
- These could be routed to a variety of controls, such as filter cutoff, filter Q, VCA volume, VCA Stereo panning control, etc.
- Because this was one of the very first MIDI based sampling keyboards, the same controls could be assigned to any MIDI controller function as well

Source : E-mu Systems, Inc.

Emulator II Floppy/Hard Disk

- Because RAM memory was limited (512K) in the early 1980's, it was important to have the sample sounds and patch information be able to be stored on disk for easy loading and sound sample distribution
- Remember this is prior to the Apple Mac and most other personal computers (no laptops!) so we had to write our own disk operating system for the Emulator Sampling Keyboards

Source : E-mu Systems, Inc.

Emulator II Internals

Source : E-mu Systems, Inc.

Current Samplers

- There have been tremendous improvements in the 30 years since the EII was introduced
- Most Samplers are now 16-24 bit linear and much higher sample rates (48khz - 256Khz)
- Most samplers now have lots of memory and others can sample directly to disk drives
- There are also now much better “pitch shift” algorithms that allow a greater sample range

Summary

- Analog Audio Synthesizers were the new instruments in the 1960's and 1970's
- These were augmented with Digital Sampling technology in the 1970's and 1980's
- Reduced technology costs enabled Digital Sampling technology to become commonplace in the 1990's and dominant in the 2000's
- There has been a resurgence of "analog" as new musicians desire that "phat" sound of the classic Moogs, Oberheims, and other gear. (Everything old is eventually "new" again!)

Thank You !

Questions?

Sources and References :

- MIDI Manufacturer's Association : http://www.midi.org
  The MIDI logos and graphics are Copyrighted Material owned by the MMA, and used here with their kind permission.
- Wikipedia : en.wikipedia.org/wiki/
- eMusician : http://www.emusician.com/news/0766/pitch-vs-frequency/146705
- E-mu Systems Inc.
- Creative Labs, Inc.
- Sequential Circuits, Inc.
- Moog Music, Inc.

Notes :