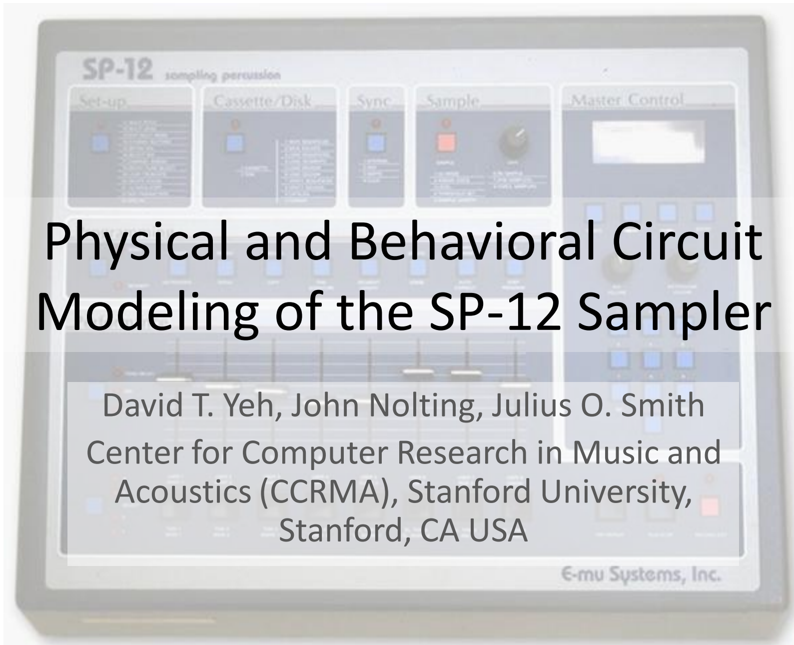

Physical and Behavioral Circuit Modeling of the SP-12 Sampler

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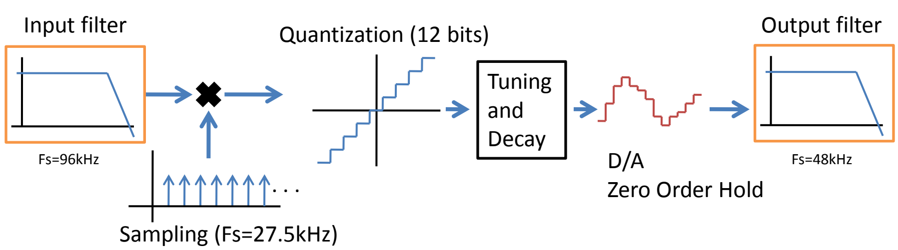
What is the SP-12

- Drum machine sampler and sequencer
- 27.5 kHz sampling rate (Fs)
- 192 kB wavetable RAM
  - Total 5 sec
  - Each continuous sound limited to 2.5 sec
- 24 internal sounds loaded from ROM
- 8 user sounds can be sampled from analog input
- 32 pitch adjustment (in half-steps), and decay envelope settings
- 8 output channels
  - Each with different equalization
  - Output 1, 2 are direct ZOH output
  - Outputs 7, 8 have dynamic filter (VCF), constant Q, exponentially swept Fc

- Often used as signal processing for drum sounds to add a “hard edge” or “grit”

Digital Model of the SP-12

- Analyze signal path of the SP-12
- Implement digitally
  - Anti-aliasing filter
  - Quantization
  - Sampling
  - Tuning and decay envelope
  - Zero Order Hold
  - Time varying filter (Voltage Controlled Filter)



Digital Model of the SP-12 examples

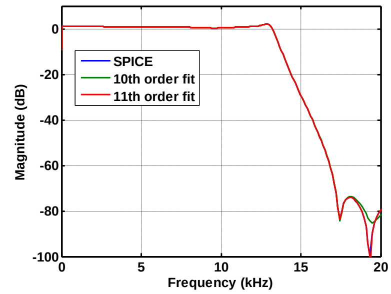
Original sample	No tune	Tune hi	Tune low	VCF
Bass Drum SP-12				
Bass Drum Matlab				
Snare Drum SP-12				
Snare Drum Matlab				
Crash SP-12				
Crash Matlab				
Ill Bass SP-12				
Ill Bass Matlab				
Bass & Guitar SP-12				
Bass & Guitar Mat				
Loop meas SP-12				
Loop meas Matlab				

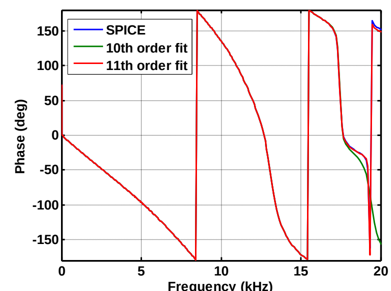
External sound examples:
Go to web table

Anti-Alias and Equalization Filters

- Fixed filters – no parameters
- Analyzed in SPICE
  - AC simulation
  - Export frequency response of input to output transfer function
- Design filters in Matlab
  - Use `invfreqz` to design digital filter
  - Increase order until good fit in band
  - Oversample at 96 kHz for easier fit

- Input anti-alias filters: 6th order
- Excellent fit through 20 kHz for input anti-alias at order 11. Lower orders may be sufficient as well.
- At Fs/2=13.75 kHz, only -10 dB attenuation -> significant aliasing

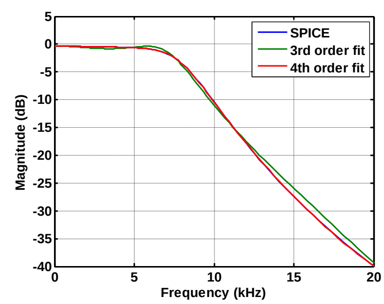


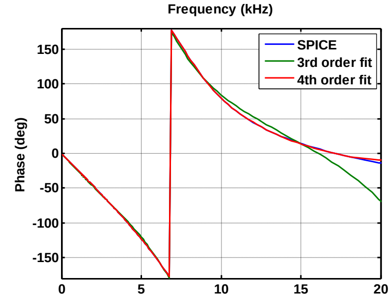


Output Equalization Filters

- Output equalization are 2 stage Sallen-Key active filters.
  - Total of 4th order.
  - Fs = 48 kHz
  - 4th order fit is good through 20 kHz.

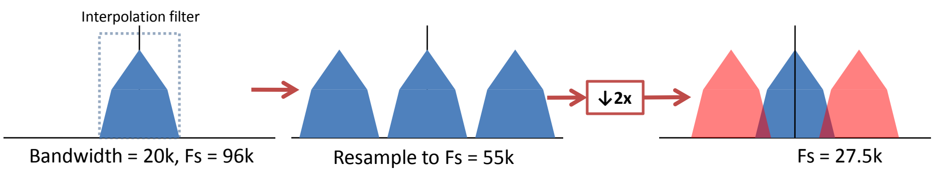
- Attenuation at Fs/2 = 13.75kHz is -23 dB
  - Incomplete rejection of spectral images





Ideal interpolation and sampling

- Signal processing rate is 96 kHz. Must convert to 27.5 kHz, and simulate the sampling process with correct aliasing.
- Interpolate to time grid of 27.5 kHz
  - Resample to multiple of 27.5 kHz for oversampling
- Downsample directly with no additional anti-alias filter
  - Previously anti-aliased by SP-12's filter



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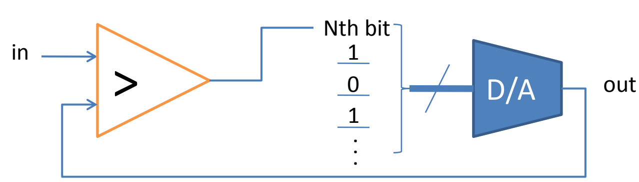
A/D conversion

- Successive approximation
  - Binary search for analog signal in digital representation
- Assumed ideal due to accuracy of process
  - Single comparator, D/A usually accurate
- 6 dB per bit, 12 bits = 72 dB SNR
- Quantization generates harmonic distortion



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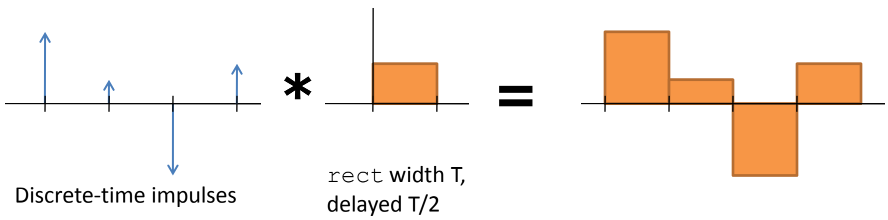
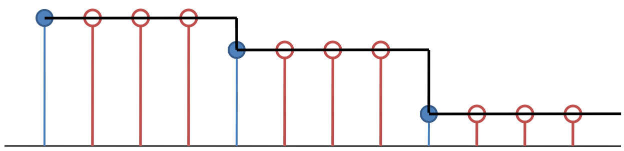
D/A conversion: Zero Order Hold



- Convolution of samples with delayed `rect`
- Staircase output – Much aliasing remains

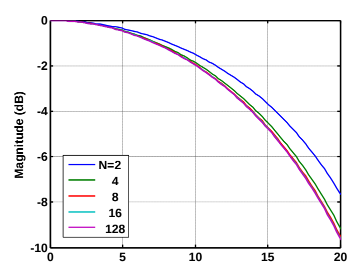
Discrete-time rect for digital implementation

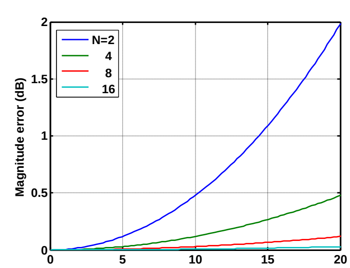


Repeat each sample by N-1 for an oversampling rate of N

Zero Order Hold

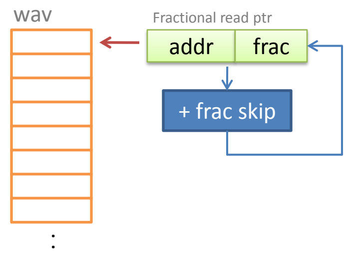
- Zero Order Hold (ZOH)
- Simple method of D/A conversion (converts impulse train to continuous time signal)
- Digitally, repeat each sample N times
- Equivalently, filtering by a frequency aliased sinc
  - Sinc plotted to 20 kHz for various discrete-time rect of width N.
- Digital ZOH has frequency domain error, but is inherently oversampled to minimize errors at lower frequencies
  - At N=4, error through 20 kHz is <0.5dB



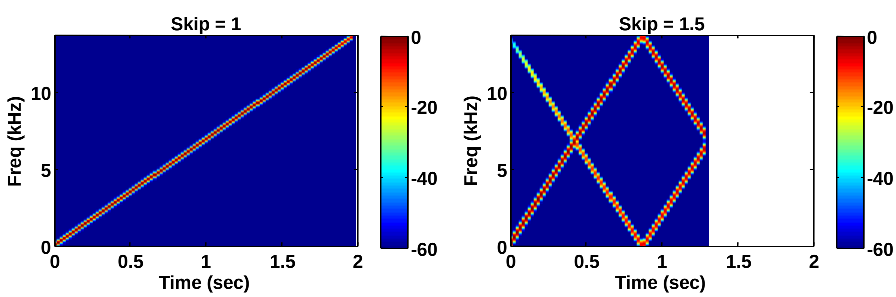


Drum tuning

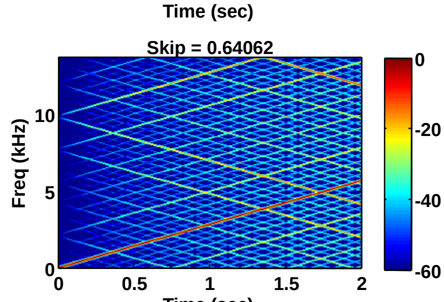
- Waveform table is held in memory
- Move read pointer by a fractional amount and round or truncate to nearest address
- Creates complicated aliasing type patterns and introduces high pitched components to sound character



Spectrograms of tuning algorithm

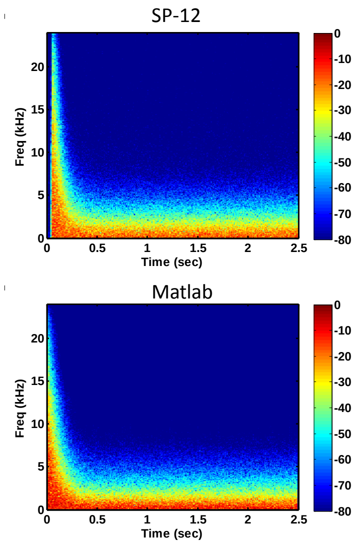


- Linear sweep from 0-20kHz.
- Fractional tunings produce various granularities of chain-link-fence-like patterns
- Tuning algorithm also changes length of waveform
- SP-12 has 32 different tuning settings with various skip amounts

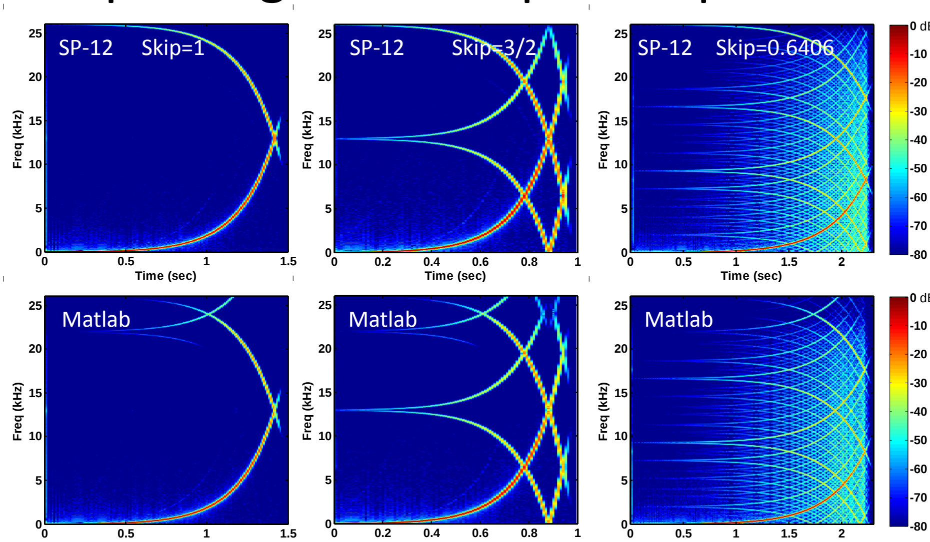


Dynamic filters

- Voltage controlled filter
  - Essentially a Moog VCF with the transistors integrated onto a single chip
  - Configured for constant Q
  - Approximately exponential cutoff frequency trajectory
- Matlab implementation
  - Digital Moog implementation (Stilson 96, Valimaki 06)
  - Exponential time constant = 0.085 s
  - Initial Fc = 14150
  - Final Fc = 1150
- Noise sweeps of SP-12 (top) and Matlab (bottom)



Spectrogram Sweeps Comparison



Sound examples

- Physical modeling produces accurate results capturing much of the character of the real thing.
- When device and schematics are available, reimplementing the functionality digitally requires little or no tweaking.