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More Than 50 Years of Artificial Reverberation

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Outline

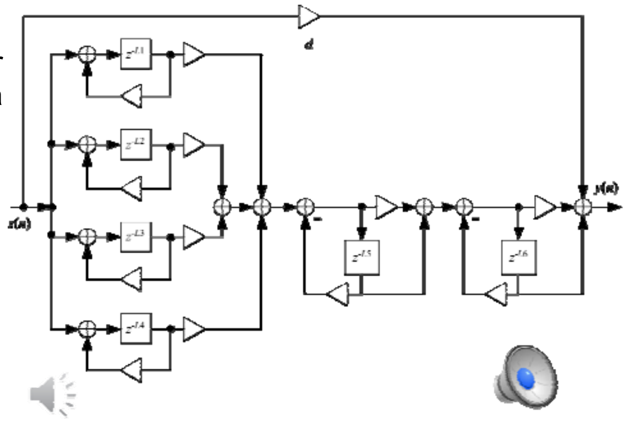
- Schroeder reverb and delay networks
- Convolution algorithms
- Velvet noise based reverberation algorithms
- Scattering delay networks
- Modal reverberation architecture

Main references for this talk:

✓ V. Välimäki, J. D. Parker, L. Savioja, J. O. Smith, and J. S. Abel, “Fifty years of artificial reverberation,” *IEEE Trans. Audio, Speech, and Language Processing*, vol. 20, no. 5, pp. 1421–1448, July 2012.
✓ V. Välimäki, J. D. Parker, L. Savioja, J. O. Smith, and J. S. Abel, “More than 50 years of artificial reverberation,” *this conference*, 2016.

Schroeder's Reverb Algorithm

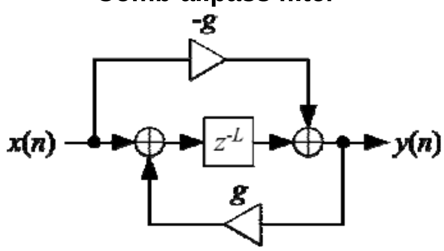
- In the early 1960s, Manfred Schroeder Introduced the idea of artificial reverb
  - Comb filters in parallel
  - Use long delay lines with incommensurate lengths
  - A few allpass filters in cascade



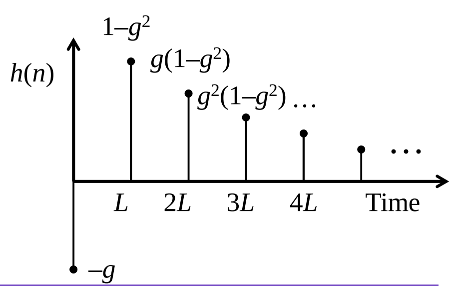
Comb-Allpass Filter

- Schroeder and Logan (1961) developed the digital allpass filter
  - Add a negative feedforward path to a comb filter
  - High echo density without spectral coloration

Comb-allpass filter

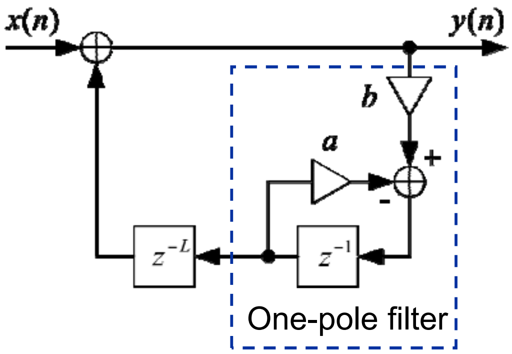


Impulse response



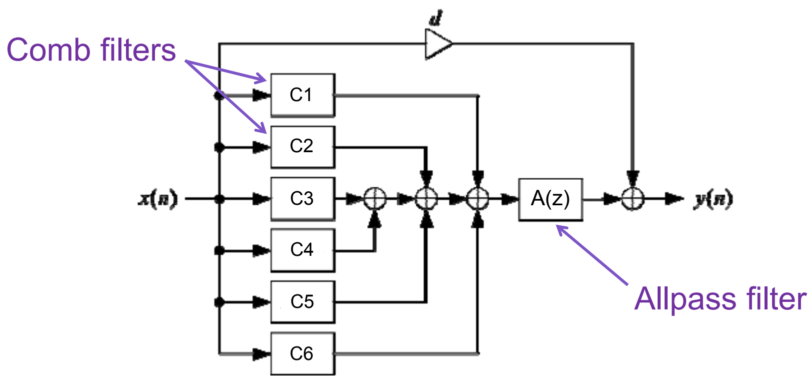
Moorer Reverb Algorithm

- Moorer (1979) inserted a one-pole lowpass filter inside the comb filter to provide frequency-dependent reverb time control
  - More natural, less metallic sound



Moorer Reverb Algorithm

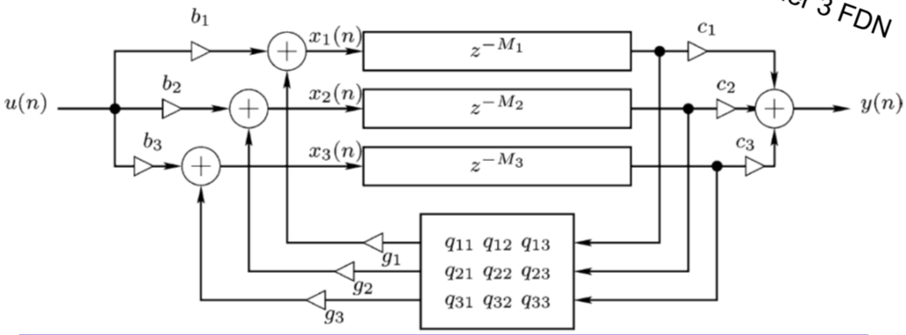
- Moorer (1979) suggested to use more comb filters and just one allpass filter



Feedback Delay Network

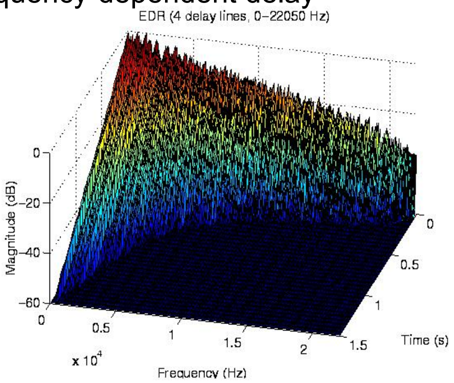
- A 'vectorized' comb filter (Jot & Chaigne,1991)
  - Based on ideas by Gerzon, Stautner & Puckette
- Lossless feedback matrix provides diffusion

Order 3 FDN



Feedback Delay Network

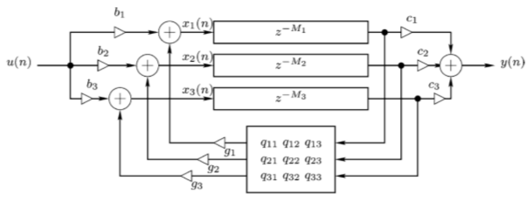
- The loop filters control the frequency-dependent delay
- Order 4 FDN sound examples
  - Original
  - Fairly dry room
    T60 = 0.7 s at 0 Hz
    T60 = 0.14 s at 22 kHz
  - Very reverberant
    T60 = 3.0 s at 0 Hz
    T60 = 0.6 s at 22 kHz



(Sound examples and figure produced by Riitta Väänänen, 2000)

FDN: Recent Developments

- How to choose the feedback matrix? How to connect the FDN to the physical world?
- Schlecht and Habets (2015) modulate the feedback matrix coefficients: modulated unitary rotation!
- Bai et al. (2015) used principles of acoustic radiance transfer and the room acoustic rendering equation



Reverberation Algorithm Categories
(Välimäki et al., 2012)

1. ***Delay networks***
   - Simulate the reverberation process using delay lines, filters, and feedback connections
2. ***Convolution algorithms***
   - Convolve a dry input signal with a recorded, approximated, or simulated room impulse response (RIR)
3. ***Computational acoustics***
   - Simulate the propagation of sound in a specified geometry
4. ***Virtual analog models***
   - Simulate electromechanical or electrical devices used formerly for producing reverberation effects (tapes, plates, springs)

Convolution Reverb

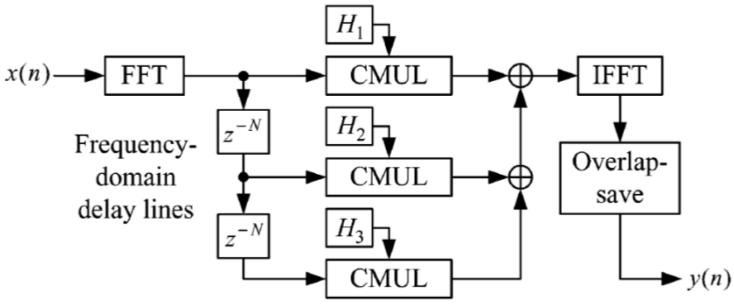
- Measure and store impulse responses of real spaces
  - A long FIR filter with RIR samples as coefficients (cf. sampling)
- A “dry” signal is filtered with the FIR filter
- RIR may have to be denoised/extrapolated
  - Can be done in time-frequency plane (Carpentier et al. 2013)
- RIR can also be obtained using numerical simulations or from a scale model (Katz et al. 2014)
- Convolution is computationally intensive
  - Example: A stereo impulse response of 2 seconds (f_s = 50 kHz)
    $\rightarrow$ 2 × 100,000 multiplications / sample
    $\rightarrow$ 10^{10} multiplications / s (= 10 GFLOPS)

Fast Convolution

- Savings can be obtained by implementing convolution by multiplying the FFTs of the signal and the filter
  - FFT is much more efficient in regular CPUs than convolution
- Basic fast convolution causes latency of 2N samples when N is RIR length
  - It takes N samples to fill the buffer, and computations are done during the next N samples (to equalize the CPU load)
  - For example, for a 2-sec RIR, the latency would be 4 seconds!

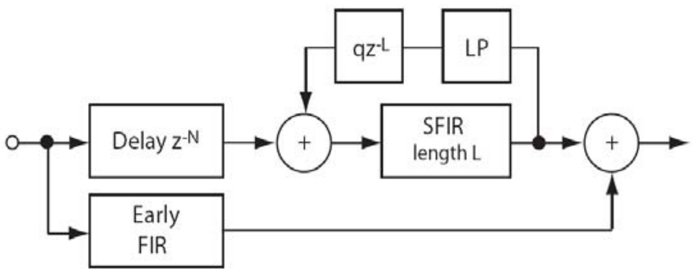
Partitioned Fast Convolution

- Kulp (1988) suggested segmenting RIR into short frames and store their FFTs (H_1, H_2, H_3 …)
  - Input signal is processed in frames of the same length
- Can be implemented with 1 FFT and 1 IFFT per frame!
  - For recent developments, see PhD thesis of Frank Wefers (2014)



Reverberators Using Sparse FIR filters

- Moorer (1979) claimed that late part of RIR is reminiscent to exponential decaying white noise
- A feedback loop containing a **sparse FIR** (SFIR) filter with random coefficients (Rubak & Johansen, 1998)
- SFIR filter coefficients can be **velvet noise** (Karjalainen & Järveläinen 2007)

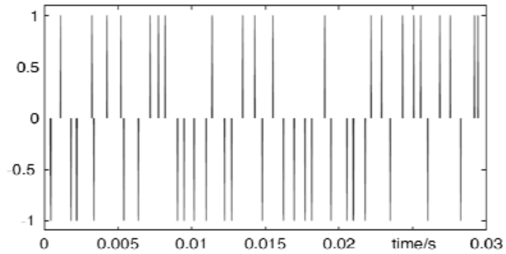


http://www.acoustics.hut.fi/demos/VelvetReverb/

Velvet Noise

- +1 or -1 samples randomly in short frames, other samples are 0
- Sounds smoother (less rough) than Gaussian noise (Karjalainen and Järveläinen, 2007; Välimäki et al. 2013)
- Signals can be convolved with 'velvet noise' efficiently without multiplications

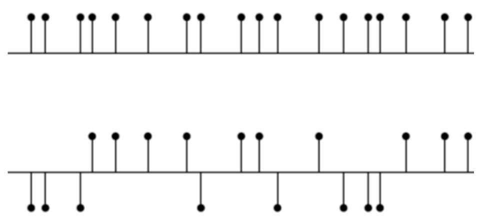
Gaussian noise   Velvet noise: 3000 spikes/sec   1000 spikes/sec   200 spikes/sec



http://www.acoustics.hut.fi/demos/VelvetReverb/

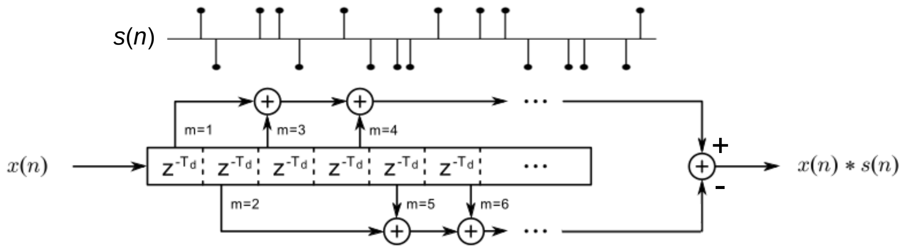
How to Make Velvet Noise

- Start with an impulse train (period T_d)
- Offset each impulse randomly within T_d
- Randomly change the sign of each impulse
- Done!



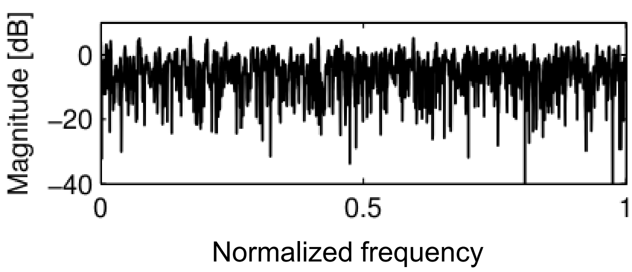
Convolution With Velvet Noise

- Use velvet noise sequence *s*(*n*) as FIR filter coefficients
- Most coefficients are **zero** and the rest are either **-1** or **+1**
- Convolution becomes simple: Add samples at +1 coefficients, subtract samples at -1 coefficients
- Sparse multiplier-free convolution!



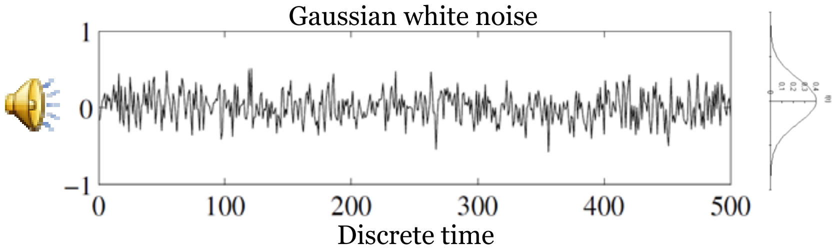
Magnitude Response of Velvet Noise FIR Filter

- Broadband response with random fluctuations
- Not exactly flat but almost



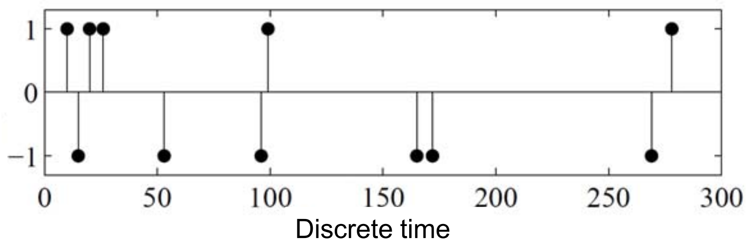
Other Noises: Gaussian White Noise

- Traditional white noise is **not sparse**
- **Gaussian noise** = random samples drawn from a normal distribution
- **Uniform** noise = random samples drawn from a rectangular distribution



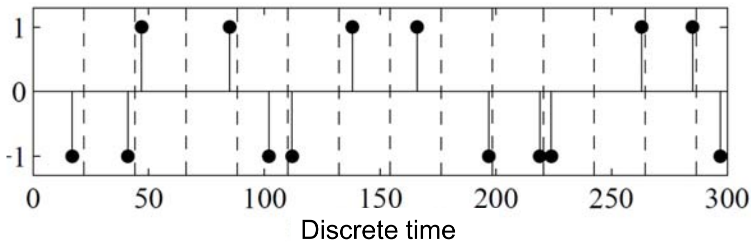
Other Noises: Totally Random Noise

- Uniformly distributed noise is scaled and rounded to values 0, -1, and +1 (Rubak & Johansen, 1998)
- **It does not sound 'smooth'**
  - Weaknesses: bursts of impulses, long gaps between impulses



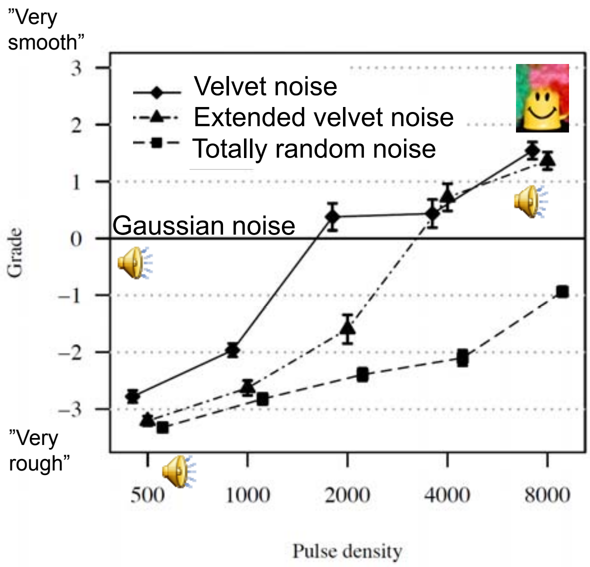
Velvet Noise

- Velvet noise fills the space evenly: no bursts, no gaps
- Divide time in frames for desired *pulse density*
  - e.g., T_d = 22 samples → 2000 impulses/s, when f_s = 44 kHz

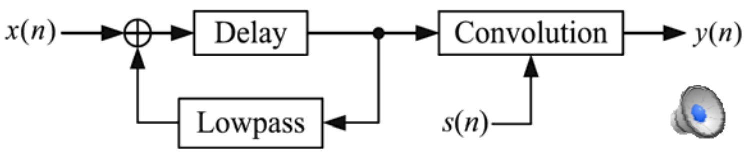


Listening Test Results

- Smoothness of ternary noises w.r.t. Gaussian white noise
  (Välimäki, Lehtonen, Takanen, 2013)
- Velvet noise is smoother than Gaussian noise, when pulse density ≥ 2000 p/s



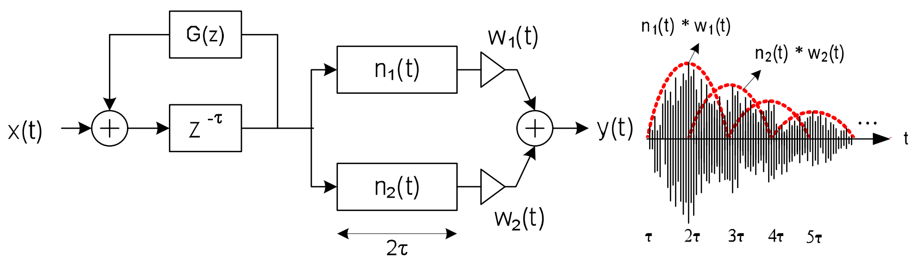
Basic Recursive Velvet Noise Reverb



- Remaining challenge: flutter echo caused by repetition

Switched Convolution Reverb

- Velvet noise reverb with low cost and memory requirements (Lee, Abel, Välimäki, Stilson, and Berners 2012)
  - Comb filter drives a time-varying velvet noise convolution
  - Velvet noise sequence is gradually updated or "switched"

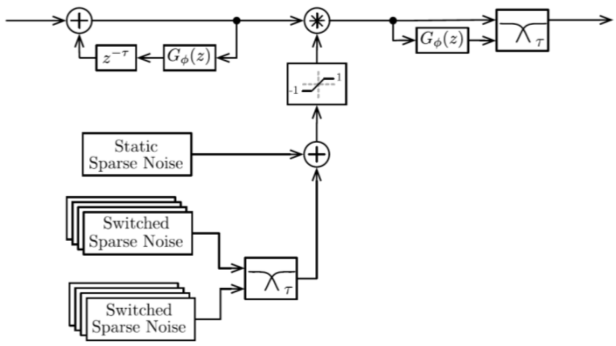


Impulse response with switching

With switching: Stationary sample

Switched Convolution Reverb '13

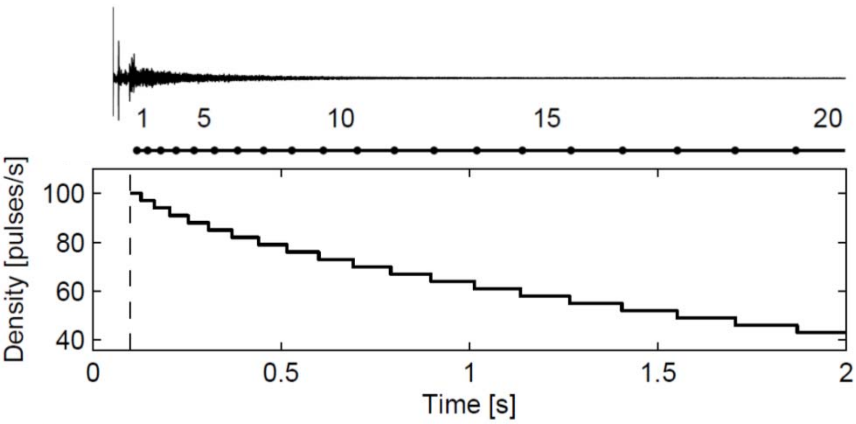
- In another version, static and crossfaded velvet noise sequences are combined (Oksanen et al. ICASSP 2013)



Medium T60 Long T60

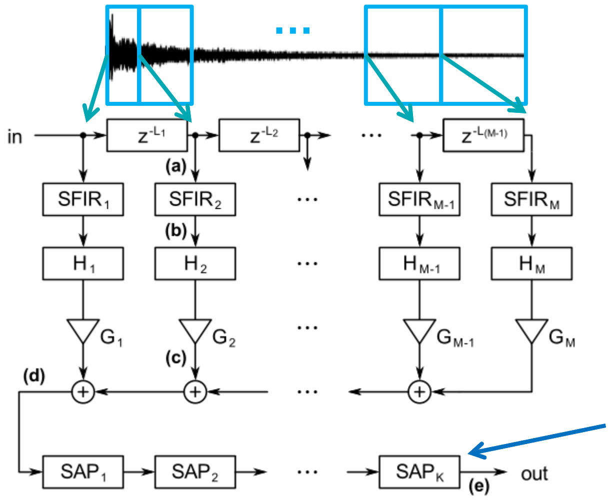
RIR Modeling Using Velvet Noise

- Measured RIR is divided into frames, first shorter, then longer
- The pulse density can gradually decrease towards the end of RIR



Ref: Holm-Rasmussen *et al.*, DAXF 2013

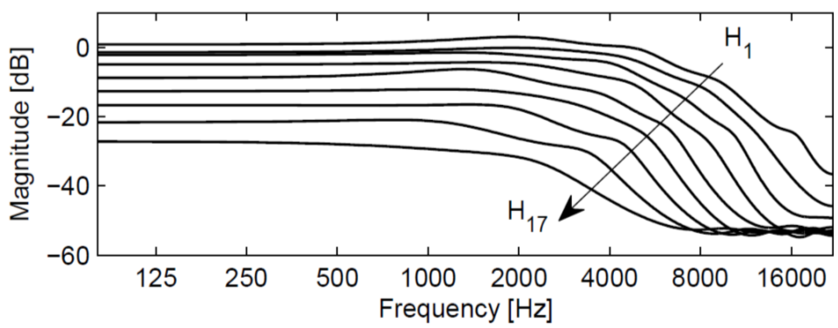
RIR Modeling Using Velvet Noise



- Delay line
- Velvet noise filters
- Lowpass filters
- Gain factors
- Add all branches
- Cascaded Schroeder all-pass filters

IIR Filter Design

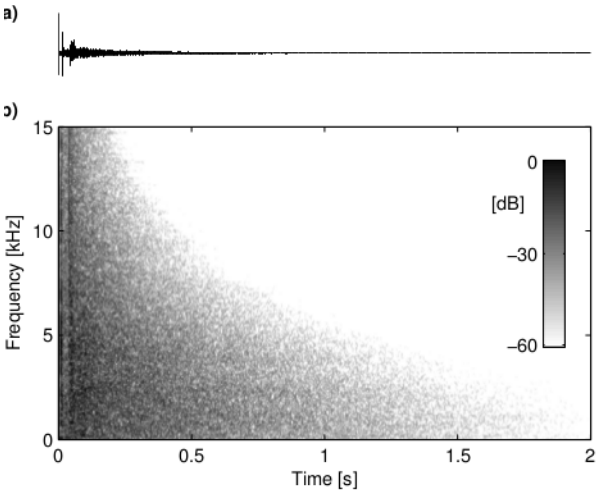
- Linear prediction of order 10 is used for estimating a filter for each frame



Example Design

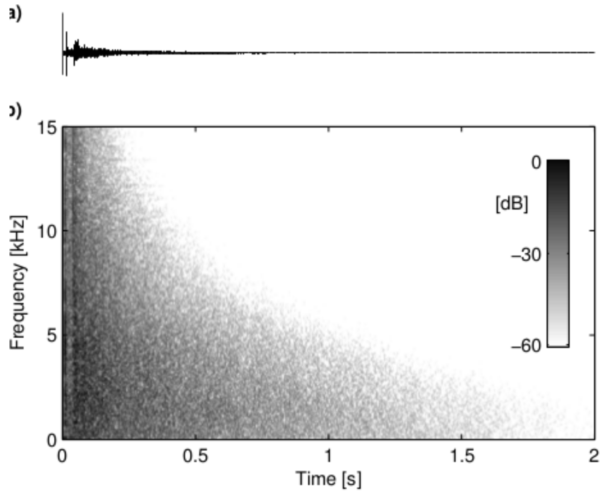
Concert hall in Pori, Finland, Merimaa et al. (2005)

Measured Impulse Response



Ref: Holm-Rasmussen
et al., DAXF 2013

Modeled Impulse Response



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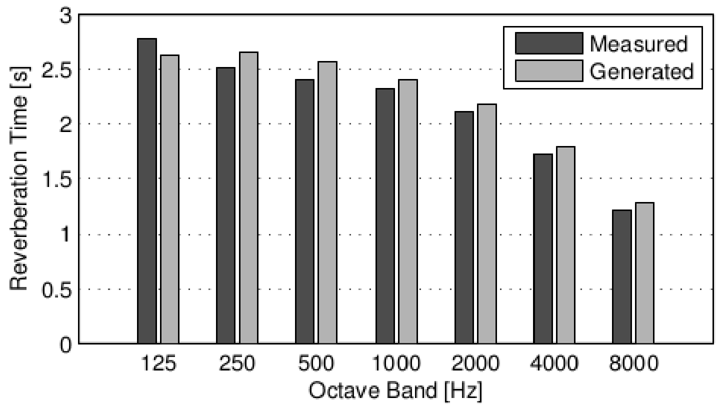
Ref: Holm-Rasmussen
et al., DAXF 2013

Comparing Measured and Modeled IR

- Close approximation of frequency-dependent reverb time is possible
- Depends on the filters and gain factors of each frame



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Ref: Holm-Rasmussen
et al., DAXF 2013

Computational Cost & Memory Usage

- Exact computational load & memory usage for a 2-sec RIR
- Proposed algorithm uses < 1% of operations of Direct Convolution; only slightly more than the best FFT-based convolution
- Proposed method has the same memory usage as Direct Convolution

No latency!

	Direct Convolution (FIR filter)	*Partitioned FFT Convolution (best so far!)*	***Proposed Algorithm***
FLOPS/sample	176,401	**399**	**627**
Delay-line memory	**88,200**	176,400	**90,442**

Ref: Wefers & Vorländer, 2012

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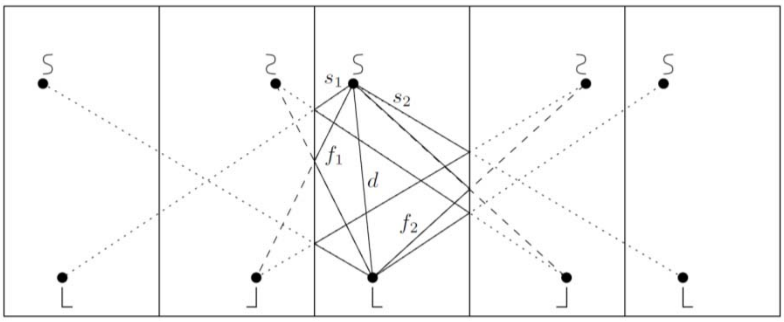
Ref: Holm-Rasmussen *et al.*, DAXF 2013

Scattering Delay Network

- New efficient way of approximating geometric ray tracing using a **digital waveguide network** (De Sena et al. 2015)
- Consider an image model of a room; side wall reflections only



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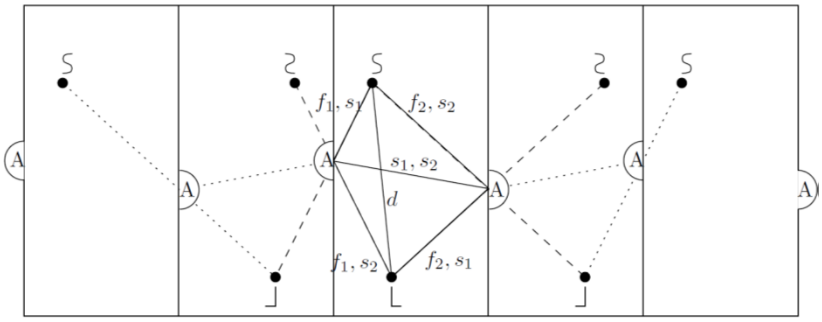
Ref: De Sena *et al.*, IEEE/ACM T-ASLP, 2015.

Scattering Delay Network (2)

- Can be approximated with digital waveguides and 2 scattering junctions (at the points of 1st-order reflections)
- 1st-order reflections are correct; 2nd-order slightly incorrect



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Ref: De Sena *et al.*, IEEE/ACM T-ASLP, 2015.

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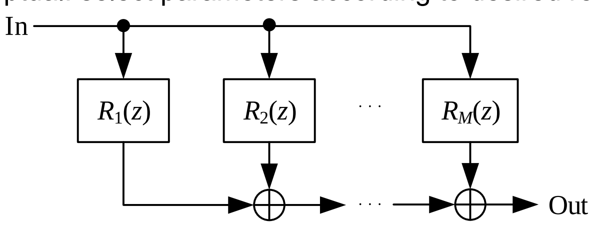
Modal Reverb Architecture

- RIR modeling in terms of its modal structure (Abel et al. 2014)
- Direct implementation as sum of many (!) parallel resonant filters
  - For each mode: amplitude, frequency & damping

1. Behavioral approach: fit mode parameters to measurements
2. Analytical: derive mode parameters from the physics
3. Perceptual: select parameters according to desired response



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Ref: Abel *et al.*, AES 137th Conv, 2014.

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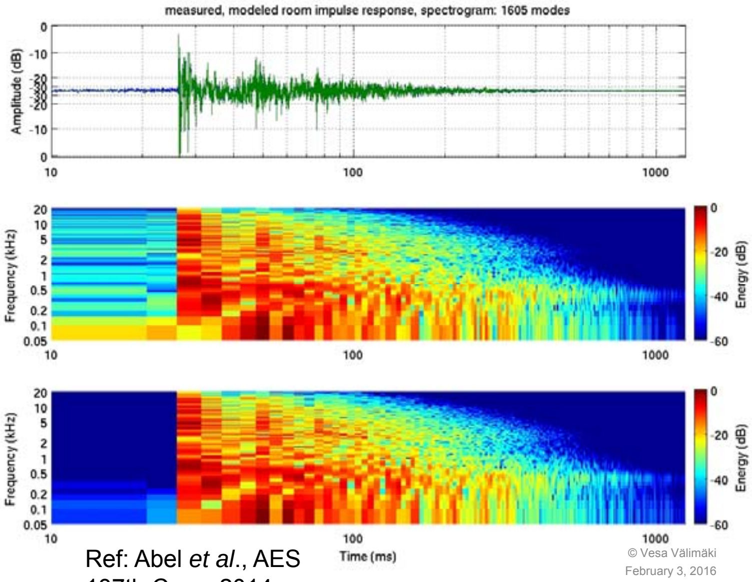
Modal Reverb Model of a Classroom IR



Ref: Abel *et al*., AES 137th Conv, 2014.

Conclusion and Future Prospects

- FDN, Schroeder, and convolution reverbs continue to be popular
- New exciting possibilities: scattering delay network, velvet noise based reverb, modal reverb architecture
  - Filtered velvet noise and modal reverb allow easy synthesis of arbitrary RIRs
- Growing interest in modeling outdoor environments and multichannel reverberation
- Artificial reverberation research is still going strong – everything has not been invented yet!

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