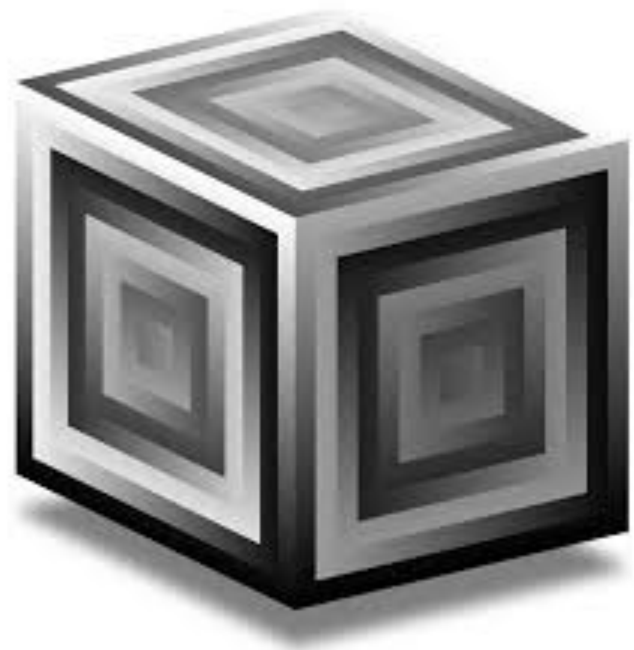
Reverb

Topics Addressed

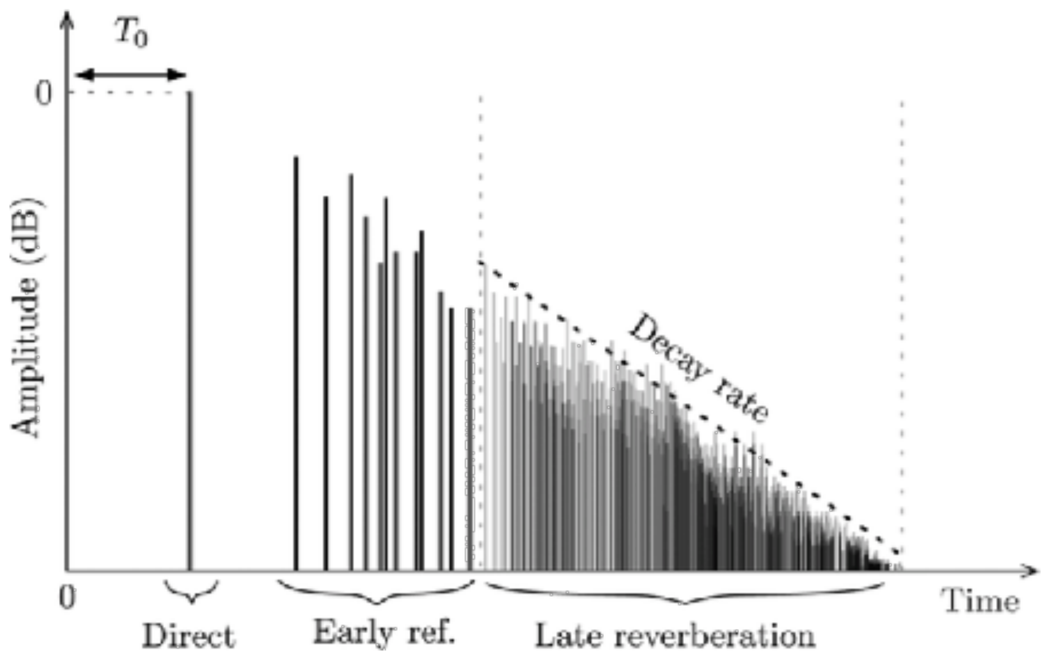
- Reverberation
- Crude Reverb
- Allpass Filter
- Schroeder Reverb
- Infinite Reverb
- Convolution Reverb



Reverberation

- Reverberation is a naturally occurring phenomenon where sound is reflected across walls and surfaces, leading to a persistence of sound after it has been produced.
  - As sound is reflected across surfaces, sound waves are absorbed and generally reflected at smaller amplitudes.
- Nearly all sound we hear has reverberance.
  - Anechoic chambers are places where no reflections are created.
- Sound generally emanates in all directions.
- Reverb is one of the most common effects in the digital audio world and is difficult to replicate naturally

Reverberation



Consider a direct sound source that is an impulse (i.e., a single sample of sound) in some reverberant space. The chart on the left shows the amplitude of reflections. Note how the early reflections are spread out and the late reflections become denser and decay in amplitude.

This chart depicts an impulse response, which details the reflections of a room/space based on a single impulse. Since sound is a sum of impulses, all moments of sound will respond in similar fashion.

Key Ideas in Reverberation

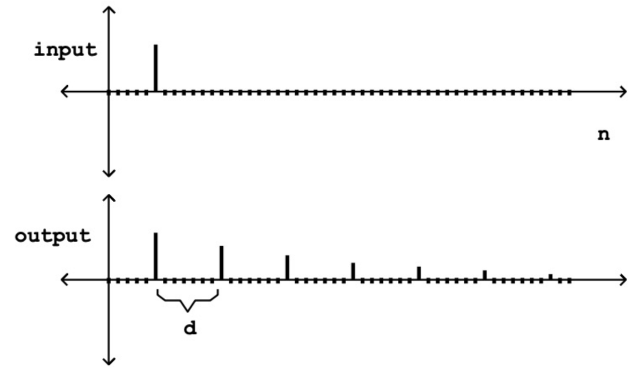
- Reverberation time – the time it takes for the reflections to reach -60db relative to the direct sound source.
- Early reflections that arrive within 50-80 milliseconds are not heard as separate from the direct sound.
  - When the early reflections are greater than 50-80ms, we perceive the reflections as an echo. This is why speaking in a canyon where the sound travels a great distance before it is reflected creates an echo.
- Pre-delay is the time between the direct sound source and the first early reflection.
  - Good indicator for the size of the room

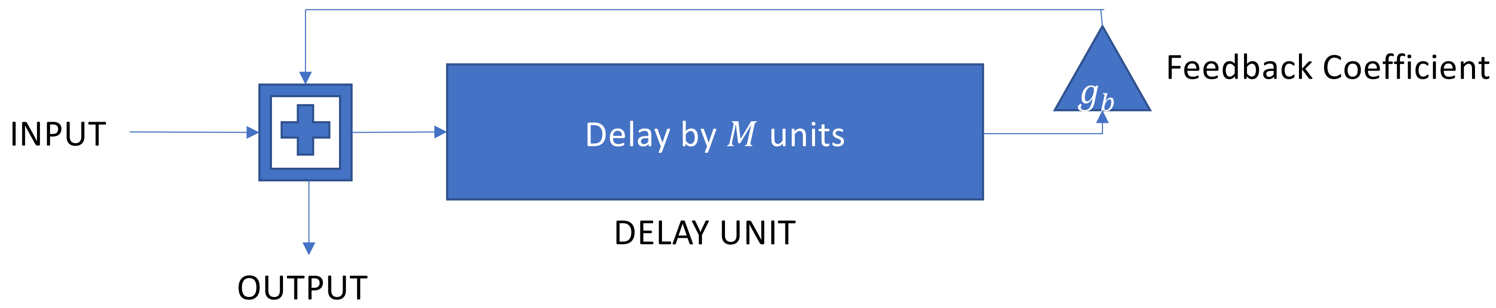
Key Ideas in Reverberation

- The absorption of reflections against surfaces is frequency dependent.
  - Higher frequencies tend to be absorbed more than lower frequencies.
  - Lower frequencies tend to have a longer reverberation time.
  - Artificial reverberation tends to use some sort of damping of high frequencies by passing the final result through a low-pass filter.
- Density of early reflections can suggest the size of the room.
- We should not hear individual echoes. The echo density should ideally be at least one thousand per second.

A Crude Reverb

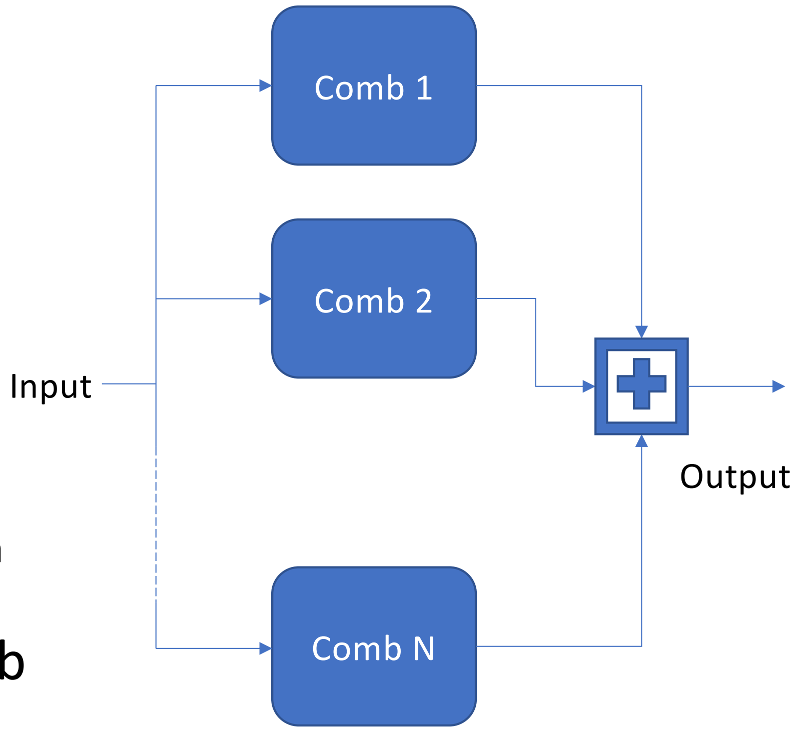
- Consider the simple feedback comb filter below that we examined earlier
- It's decaying and infinite response mirrors the kind of impulse response similar in reverberation.
- How can we use the feedback comb filter to construct a reverb?





In Parallel

- If we place the comb filters in parallel, how many filters would we need to achieve a good enough echo density?
  - At least about 40!
  - The delay for each one should be unique and randomized to avoid accentuating certain frequencies. This will give us a mostly flat amplitude response, which is a plus!
    - Remember individual feedback comb filters accentuate certain frequencies. They do not have a flat amplitude response.
- Efficient? Nope. We are using a lot of comb filters!

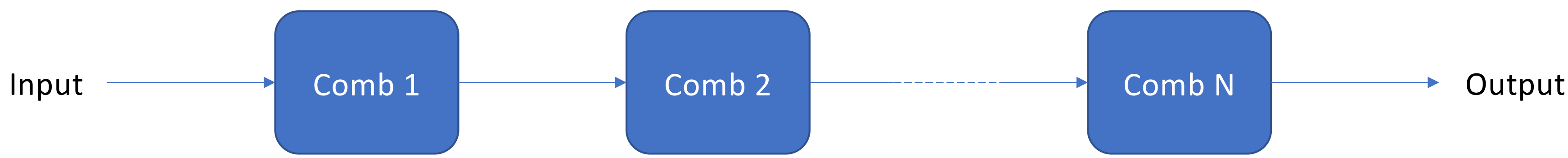


In Parallel

```
(
SynthDef(\parallelCombReverbStereo, {
	arg in, out = 0, maxdelaytime = 0.2, decaytime = 1, inputGain = 0.5, reverbGain = 0.2;
	var reverb, input, delayTimes, numOfCombs = 40;
	input = In.ar(in, 2);
	reverb = 0;
	numOfCombs.do({
		arg delayTime;
		reverb = reverb + CombN.ar(
			input,
			maxdelaytime,
			Rand(0, 0.02).asStream.next, // Randomized delay time
			decaytime,
			1/numOfCombs // Amplitude
		)
	});
	Out.ar(out, (input * inputGain) + (reverb * reverbGain));
}).add;
)
```

In Series

- If we place the comb filters in series, we get a different effect.
- It only takes roughly four or five filters to achieve the desired echo density
  - Each single impulse’s repetitions are repeated exponentially.
- Amplitude response is **not** flat and certain tones will be brought out of the reverb. See the coding example to hear the issue. This is not desirable. The parallel version produces a better reverb even if it is less efficient.



In Series

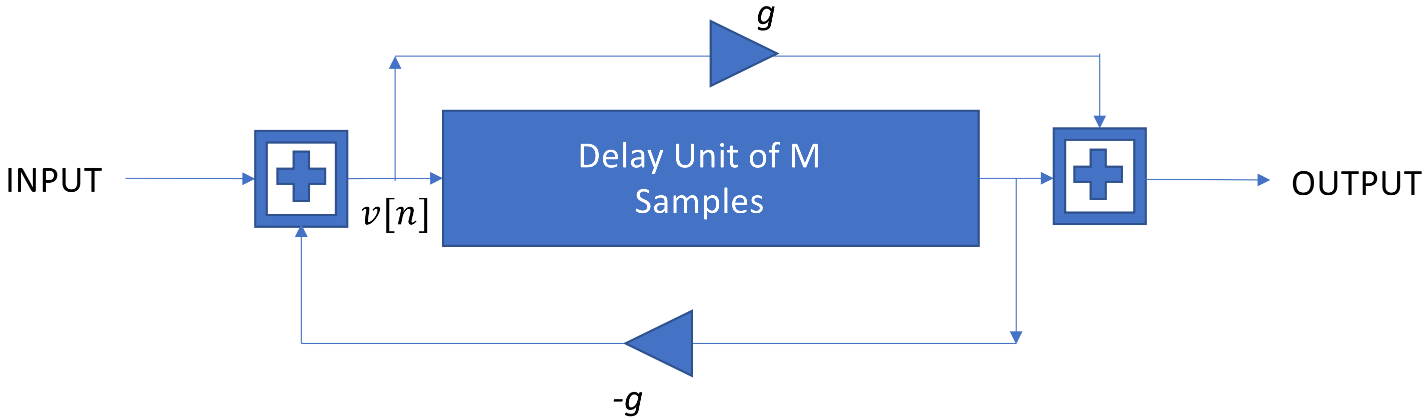
```
(
SynthDef(\seriesCombReverbStereo, {
	arg in, out = 0, maxdelaytime = 0.2, decaytime = 2, inputGain = 0.5, reverbGain = 0.9;
	var reverb, input;
	input = In.ar(in, 2);
	reverb = input;
	// Successively smaller delays to attempt to mirror the late reflection times
	[0.03, 0.0221, 0.0114, 0.006, 0.0041].do({
		arg delaytime;
		reverb = CombN.ar(reverb, maxdelaytime, delaytime, decaytime, delaytime * 30);
	});
	Out.ar(out, (input * inputGain) + (reverb * reverbGain));
}).add;
)
```

Not very good at all!

Schroeder Reverb

- In 1962, Manfred Schroeder published an important paper on several algorithms for natural sounding reverberation.
  - He attempted to solve two problems
    - Creating a reverb with a flat amplitude response. A non-flat response leads to “coloration”.
    - Achieving a suitable echo density of >1000 impulses per second. Anything lower would lead to “flutter” and an unrealistic reverberation.
  - Identified the need for an allpass delay (i.e., an allpass filter) that would have a flat amplitude response in IIR filter (i.e., one with feedback).
    - Would prevent the issue of coloration
- His algorithm required a bank of parallel comb filters, as we have just discussed, as well as a series of allpass filters.
- Schroeder’s reverb has a more metallic sound than we desire today.

Allpass Filter



- The allpass filter is a combination of feedforward and feedback comb filter in series.
- The difference equation given an input $x[n]$, an output $y[n]$, and an intermediate step of $v[n]$ can be summarized in two equations:

$$v[n] = x[n] - g * v[n-M]$$

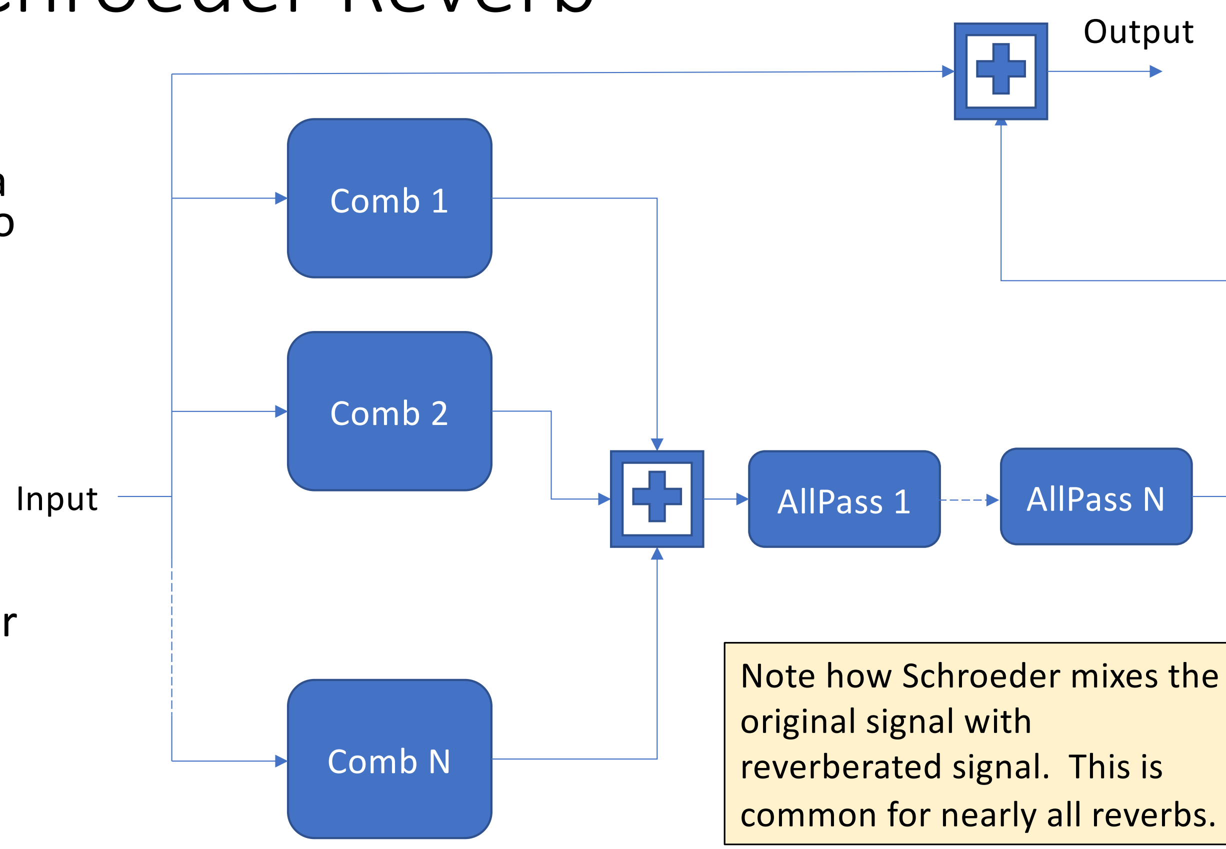
$$y[n] = g * v[n] + v[n-M]$$

Key Aspects of All Pass Filter

- Allpass filters are “colorless”. The amplitude response of an allpass filter is 1 for all frequencies so none are attenuated/amplified.
- Allpass filters do alter the phase response of the input signal.
  - Useful for phase rotation and phasers
  - Useful for smearing transients
- Allpass filters are key components of artificial reverberation due to their colorless quality.

Schroeder Reverb

- Schroeder's basic design for a natural sounding reverb was a bank of parallel comb filters to mimic the early reflections
- A series of all pass filters are used to add echo density and mimic the late reflections
- See his paper "Natural Sounding Artificial Reverberation" for more details about his specific implementation. He used four parallel comb filters and two allpass filters and he makes specific markings on the settings for each.



Note how Schroeder mixes the original signal with reverberated signal. This is common for nearly all reverbs.

Aside: Mix in SC

- The unit generator `Mix` can be used to mix down an array of signals into a single channel by adding the various signals together. We will use this to create the parallel comb filter bank.
  - Note that we could have used this UGen with our previous parallel comb filter bank.
- Particularly useful is the class method .fill
  - 1st argument is the number of signals to mixdown into a single channel
  - 2nd argument is a function expressing the signal that will be copied n times (n is given by the first argument)
    - Most of the time we will employ randomization in the signal function for the 2nd argument so that each individual signal is slightly different. This allows us to create any n number of parallel comb filters but with randomized delay times for each, for example.

Schroeder Reverb in SC

```
(
SynthDef(\schroederReverbMonoInput, {
	arg in, out = 0, reverbGain = 0.2;
	var input, predelay, earlyRef, lateRef;

	// Mono - can't modulate numChannels so would need separate UGen for stereo
	input = In.ar(in, 1);

	// Adding predelay can help create a more realistic reverberance
	predelay = DelayN.ar(input, 0.048);

	// Early reflection comb filtering which 8 parallel comb filters
	// Mix adds all the signals together
	earlyRef = Mix.fill(7, {CombL.ar(predelay, 0.1, Rand(0.02, 0.08), 3)});

	// Create a series of 4 AllPass Filters for late reflections
	lateRef = earlyRef;
	4.do({lateRef = AllpassN.ar(lateRef, 0.050, [0.050.rand, 0.050.rand], 0.7)}); // Now Stereo

	Out.ar(out, input + (reverbGain * lateRef));
}).add;
)
```

”Eerier” reverb

- We can modify the Schroeder reverb on the previous slide, by modulating the delay of the parallel comb filter bank.
- The unit generator `LFNoise` and `LFNoise1` (the latter is linearly interpolated) can generate random values between 0 and +1
- Try switching `Rand(0.02, 0.08)` with `LFNoise1.kr(0.1.rand, 0.04, 0.05)`
- This will create a sliding effect that can produce a spooky/unnatural reverb as parts of the signal receive changing delay times in the comb filtering stage.

Infinite Reverb

- The Schroeder Reverb attempts to create a natural sounding reverb. Let's try and make something more unnatural but nevertheless sonically interesting.
- Consider a series of all pass filters that feedback upon themselves.
  - Schroeder alludes to such a reverb in "Natural Sounding Artifical Reverberation" but notes that such a system would produce a non-exponential decay
    - James A. Moorer's ideal late reverberation was exponentially decaying white noise
- SuperCollider offers the UGens `LocalIn` and `LocalOut` to create a feedback loop within a SynthDef.
  - Note: only one LocalIn/LocalOut may be used per SynthDef.

Infinite Reverb

```
SynthDef(\infiniteHoldReverb, {
	arg feedback = 1, inputGain = 0.8, reverbGain = 0.3;
	var sig, local;

	// Input signal
	sig = SoundIn.ar(0); // Mono input
	local = LocalIn.ar(1) + sig;

	// Apply Reverb
	15.do{local = AllpassN.ar(local, 0.06, Rand(0.001, 0.06), 3)};

	// Feedback loop
	LocalOut.ar(local*feedback); // Consider adding a Low Pass Filter here

	//Output
	Out.ar(0, (inputGain * sig) + (reverbGain * local) ! 2); // Stereo Output
}).add;
```

Credit to Eli Fieldsteel

Infinite Reverb

- When the feedback gain is set to 1, the reverb is truly infinite and will continue to loop continuously.
  - When the feedback <1, the reverb will decay. Setting the feedback to something close to 1 will create a long decay and could be advantageous to continually add new sounds. Keeping the reverb set at 1 could lead to clipping.
  - When the feedback >1, the reverb will increase and gain and will lead to clipping and distortion. You should not set it above one.
- One could consider adding a Low Pass Filter at the LocalOut stage.
  - Most higher frequencies tend to be absorbed upon reflection and so digitally filtering out higher frequencies can lead to a more “realistic” sound.

Convolution Reverb

- Digital/algorithmic attempts at natural reverb are difficult to create.
  - Many academic papers discussing techniques/strategies
  - Many commercial digital reverbs are proprietary – don't know how they are created
- One idea: attempt to capture the natural reverb of surrounding environment (i.e., concert hall, church, room… etc.)
  - Impulse response of environment
    - Impulse
    - Sine Sweep
- Convolve the impulse response of the environment with a digital audio signal to create a natural sounding reverb.
  - See the unit generator `Convolution`