Schroeder Reverberators

The subject of *artificial reverberation* was initiated in the early 1960s by Manfred Schroeder and Ben Logan [420,421,415]. Early Schroeder reverberators consisted of the following elements [415]:

- A series connection of several *allpass filters*
- A parallel bank of *feedback comb filters*
- A *mixing matrix*

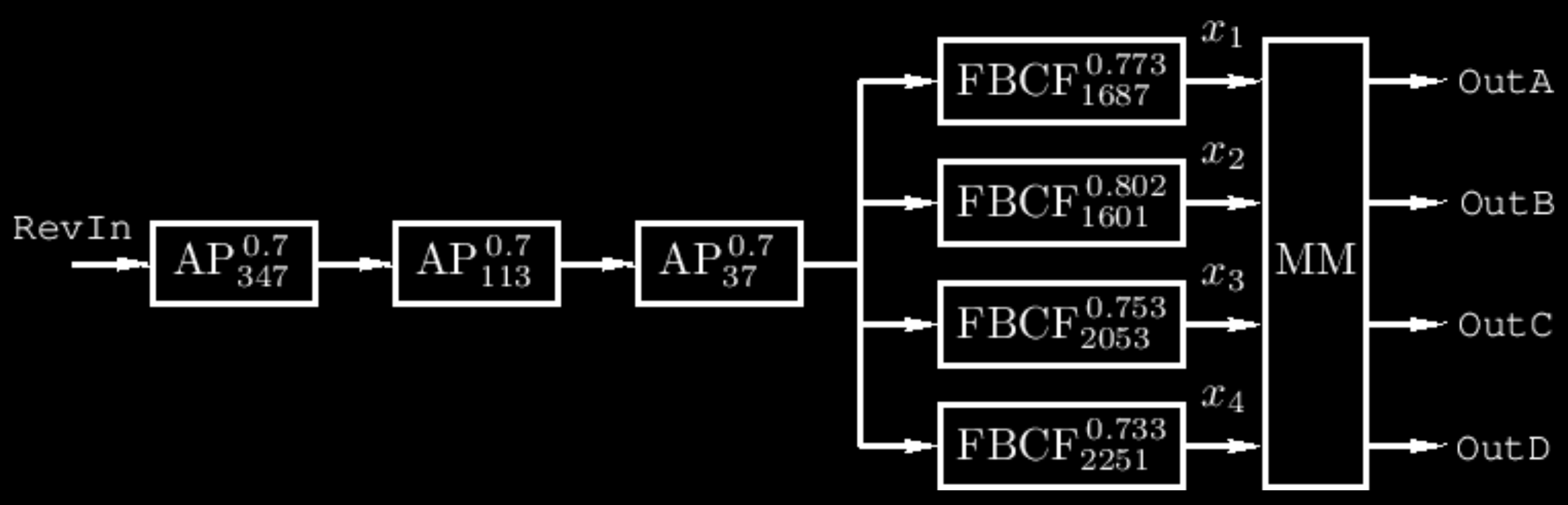

Figure: A Schroeder reverberator we will call JCREV developed by Prof. John Chowning, founding director of CCRMA (drawn from a 1972 MUS10 software listing, where MUS10 was an ``acoustic compiler'' language descended from Music V [308]).

An example is shown in Fig.3.5, where, in that figure,

$$\mathrm{AP}_N^g \;\isdef\; \frac{-g+z^{-N}}{1-gz^{-N}} \tag{4.2}$$

denotes a *Schroeder allpass* section with delay length N samples and coefficients g (see Fig.2.30 and associated discussion),

$$\mathrm{FBCF}_N^g \;\isdef\; \frac{1}{1-g\,z^{-N}} \tag{4.3}$$

denotes a *feedback comb* filter with delay length N and coefficient g (diagrammed in Fig.2.24), and MM in Fig.3.5 denotes the *mixing matrix*

$$\texttt{MM} = \left[\begin{array}{rrrr} 1 & 1 & 1 & 1\\ -1 & -1 & -1 & -1\\ -1 & 1 & -1 & 1\\ 1 & -1 & 1 & -1 \end{array}\right]$$

which can be efficiently implemented using four adders and two negations:

$$\begin{aligned} \texttt{OutA} &= s_1 + s_2\\ \texttt{OutB} &= -\texttt{OutA}\\ \texttt{OutC} &= -\texttt{OutD}\\ \texttt{OutD} &= s_1 - s_2\\[10pt] s_1 &= x_1 + x_3\\ s_2 &= x_2 + x_4 \end{aligned}$$

As discussed above in §3.4.2, the allpass filters provide ``colorless'' high-density echoes in the late impulse response of the reverberator [420,421]. These allpass filters may also be referred to as *diffusers*. While allpass filters are ``colorless'' in theory, perceptually, their impulse responses are only colorless when they are extremely short (less than 10 ms or so). Longer allpass impulse responses sound similar to feedback comb-filters. For steady-state tones, however, such as sinusoids, the allpass property gives the same gain at every frequency, unlike comb filters.

Schroeder [415, p. 221] suggests a progression of allpass delay-line lengths close to

$$M_i T \approx \frac{100\ \mathrm{ms}}{3^i}, \quad i=0,1,2,3,4,$$

and chosen to be mutually prime (no common factors). The 100 ms value was chosen so that when $g=0.708$ in Eq.(3.2), the time to decay 60 dB (t_{60}) would be 2 seconds. Thus, for $i=0$, $t_{60}\approx 2$, and each successive allpass has an impulse-response duration that is about a third of the previous one. Using 5 series allpasses in this way yields an impulse-response echo density of about 810 per second, which is close to the desired thousand per second [415, p. 221].

The parallel comb-filter bank is intended to give a psychoacoustically appropriate fluctuation in the reverberator frequency response. As discussed in Chapter 2 (§2.6.2), a feedback comb filter can simulate a pair of parallel walls, so one could choose the delay-line length in each comb filter to be the number of samples it takes for a plane wave to propagate from one wall to the opposite wall and back. However, in his original paper [415], Schroeder describes a more psychoacoustically motivated approach:

> ``There are about 15 large response peaks in every 100 cps [Hz] interval for a room with 1 sec reverberation time. Thus, one might hope that if an artificial reverberator has a comparable number of response peaks it might sound just as good as a real room. We have been able to confirm this expectation by subjective evaluations of the responses of reverberators consisting of several comb filters ... connected in parallel. For a delay of 0.04 sec, the number of response peaks per 100 cps [Hz] is 4. Thus, between 3 and 4 comb filters in parallel ... with incommensurate delays, are required to approximate the number of peaks in the frequency reponse of a room having a reverberation time of T[60] = 1 sec. Also, the open loop gain of the comb filters should not exceed about 0.85 or -1.4 dB to keep the response fluctuations from being excessive.''

Thus, one may choose the comb-filter delay-line lengths more or less arbitrarily, and then use enough of them in parallel (with mutually prime delay-line lengths) to achieve a perceptually adequate fluctuation density in the frequency-response magnitude. In [415], four such delays are chosen between 30 and 45 ms, and the corresponding feedback coefficients g_i are set to give the desired overall decay time.

The delay lengths shown in Fig.3.7 were optimized by ear by John Chowning (and perhaps others at CCRMA) for an audio sampling rate of $f_s=25$ kHz.

Finally, for multichannel listening, Schroeder suggested [415] a *mixing matrix* at the reverberator output. The goal of the mixing matrix is to bring out any number of uncorrelated audio channels of reverberation (for any number of output speakers) [154, p. 111-112].

Subsections

``Physical Audio Signal Processing'', by Julius O. Smith III, W3K Publishing, 2010, ISBN 978-0-9745607-2-4.

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