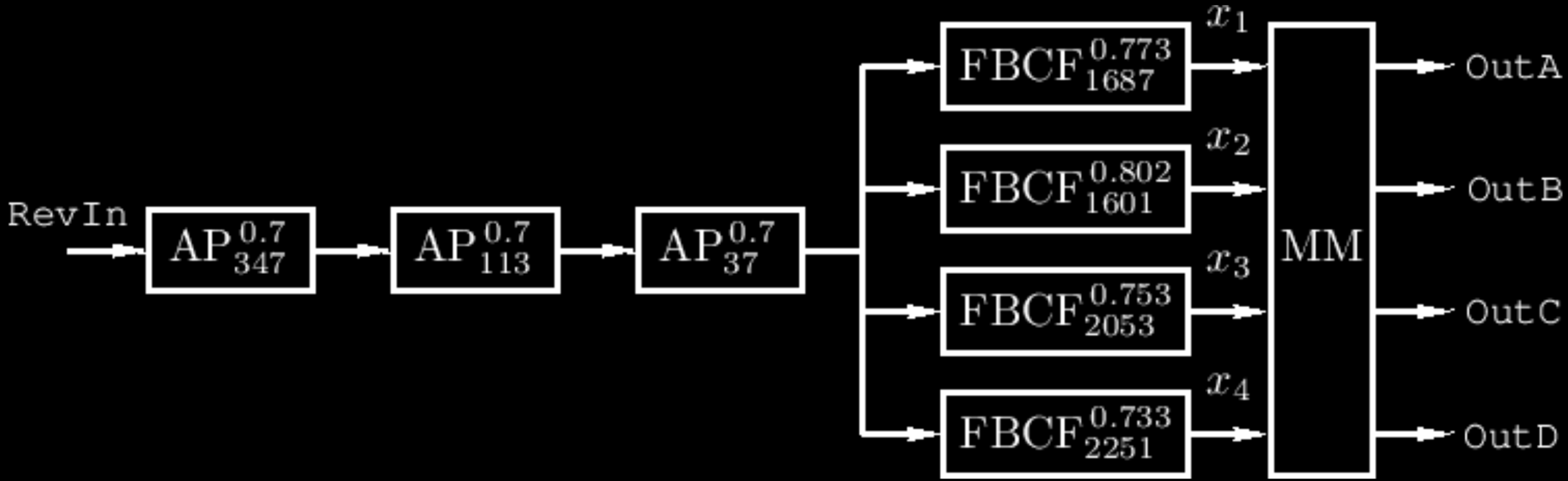


## Schroeder Reverberators

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,

$$\text{AP}_N^g \triangleq \frac{-g + z^{-N}}{1 - g z^{-N}} \tag{4.2}$$

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

$$\text{FBCF}_N^g \triangleq \frac{1}{1 - g z^{-N}} \tag{4.3}$$

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*

$$\text{MM} = \begin{bmatrix} 1 & 1 & 1 & 1 \\ -1 & -1 & -1 & -1 \\ -1 & 1 & -1 & 1 \\ 1 & -1 & 1 & -1 \end{bmatrix}$$

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

$$\begin{aligned} \text{OutA} &= s_1 + s_2 \\ \text{OutB} &= -\text{OutA} \\ \text{OutC} &= -\text{OutD} \\ \text{OutD} &= s_1 - s_2 \end{aligned}$$

$$\begin{aligned} s_1 &= x_1 + x_3 \\ s_2 &= x_2 + x_4 \end{aligned}$$

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

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

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

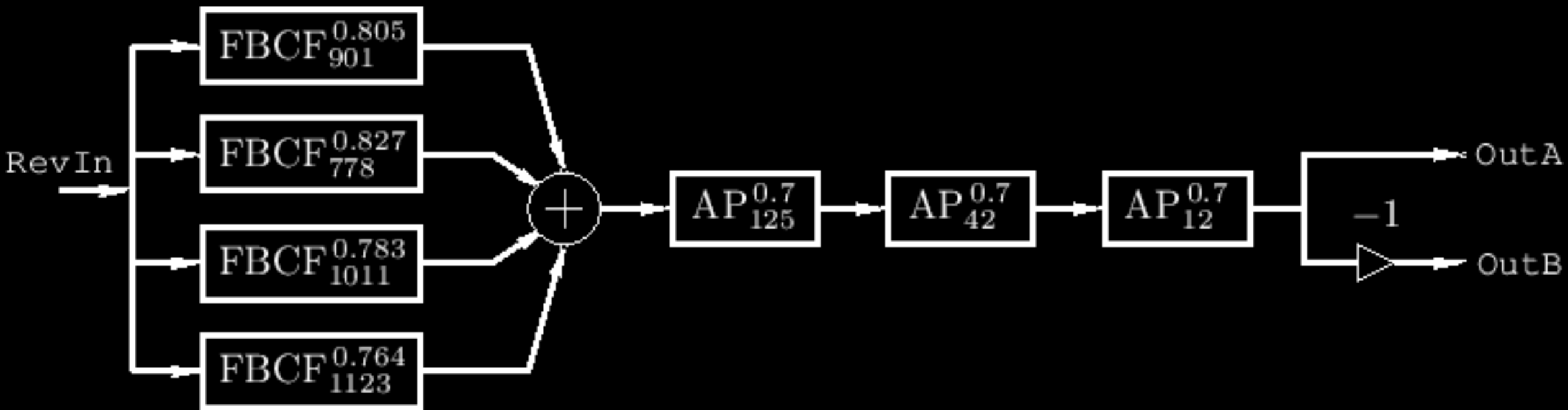
- Example Schroeder Reverberators



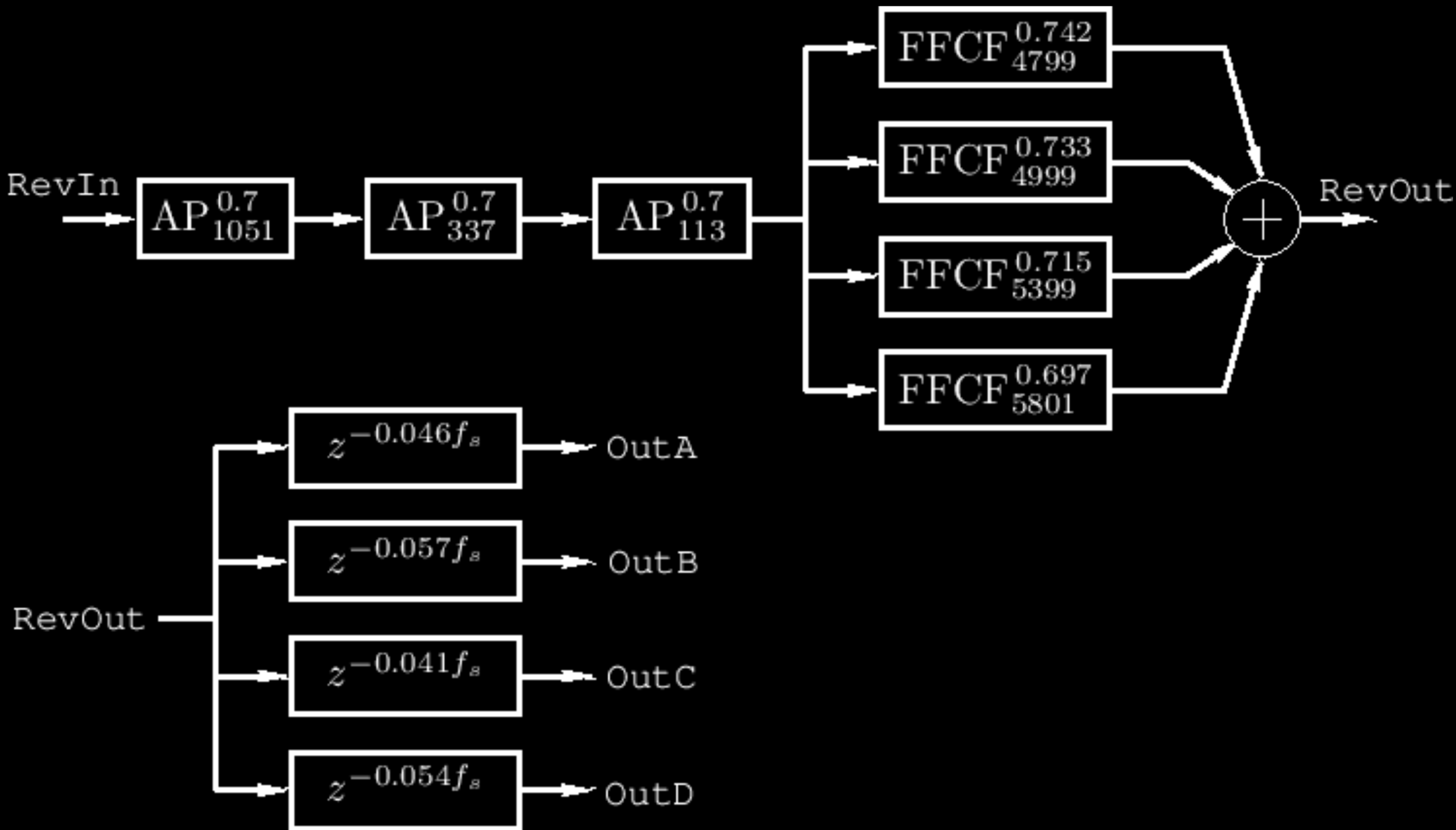


### Example Schroeder Reverberators

Additional example Schroeder Reverberators, drawn from CCRMA software listings, are shown in Figures 3.6 and 3.7. The notation used in the figures is explained above in Equations (3.2-3.3).



**Figure:** Schroeder reverberator SATREV by Prof. John Chowning at CCRMA (drawn from a 1971 MUS10 software listing).



**Figure 3.7:** A later Schroeder reverberator for the Samson Box [396] at CCRMA, based on (and still called) JCREV.

A software musical instrument using one of these reverberators simply adds its output, suitably scaled, to the real-time variable RevIn (the global reverberator input sample at the current time). In both examples, we again see three Schroeder allpass filters in series (Schroeder suggested five, as disussed above).

The rather small reverberator of Fig.3.6 is thought to have been used in John Chowning's often-heard FM-brass canon sound examples.<sup>4.8</sup> Like the more computationally expensive four-channel-audio example in Fig.3.5, it is designed for a 25 kHz sampling rate. Its ``mixing matrix" is simply a negation of the right stereo channel.

In addition to the allpass chain in Figures 3.6 and 3.7, there is a parallel bank of four feedback comb filters. Since all of the filters are linear and time-invariant [452], the series allpass chain can go either before or after the parallel comb-filter bank. Unlike the allpass filters, the comb filters have an irregular magnitude frequency response, and they can be considered a simulation of four specific echo sequences. The delay lengths in these comb filters may be used to adjust the illusion of ``room size", although if they are shortened, there should be more of them in parallel, according to Schroeder's quote above.

In Fig.3.7, the reverbator output signal RevOut is fed to the four audio output channels via four delay lines. These delay lengths are specified relative to the sampling rate  $f_s$ .<sup>4.9</sup> Output delay lines can substitute for or supplement a mixing matrix as a means for *decorrelating* the reverberation output channels (to minimize reverberation *imaging* between speakers). In this particular case, however, the delays are evidently not optimized for decorrelation. Also, for purposes of decorrelation, the shortest delay can be subtracted from the other three and its corresponding delay-line eliminated.

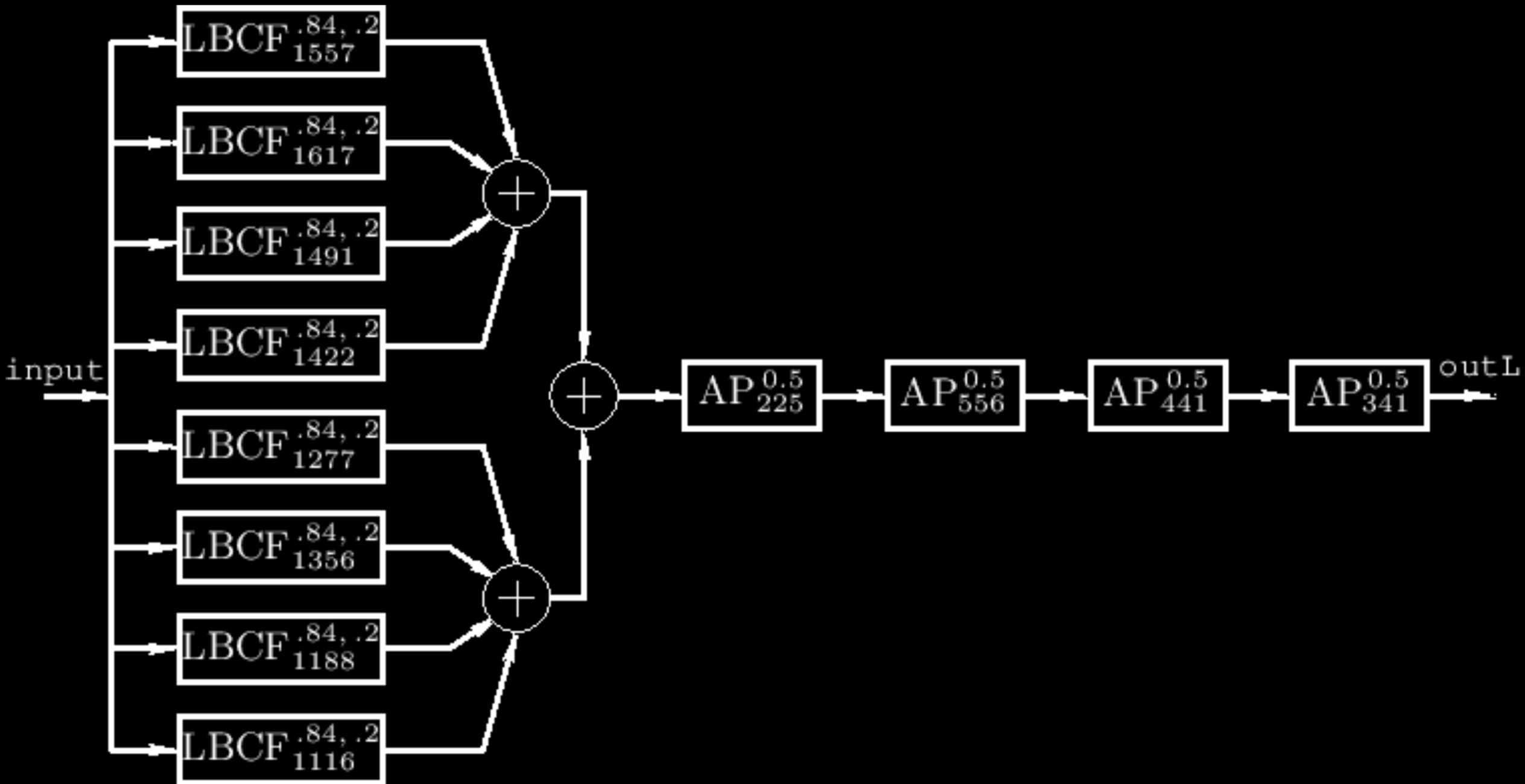
A Schroeder reverberator along the above lines may be found in the Synthesis Tool Kit (STK) [86]. See files JCREV.cpp and JCREV.h.

The reverberators shown in Figures 3.5 and 3.6 above are included in the Faust distribution. See functions jcrev, satrev, and reverb\_demo in the effect.lib library. While these Schroeder reverberators are quite small by today's standards, they are well tuned for their size. More commonly used today is freeverb (included in the examples directory of the Faust distribution as freeverb.dsp), discussed next.



## Freeverb

A more recently developed Schroeder reverberator is ``Freeverb" -- a public domain C++ program by ``Jezar at Dreampoint" used extensively in the free-software world. It uses four Schroeder allpasses in series and eight parallel Schroeder-Moorer filtered-feedback comb-filters (§2.6.5) for each audio channel, and is said to be especially well tuned.



**Figure:** Freeverb block diagram (left stereo channel). The Schroeder-Moorer lowpass-feedback-comb-filter, denoted  $\text{LBCF}_N^{f,d}$  in the figure, is defined in §3.6.2 below. The use of three summers instead of one is for drawing convenience.

Figure 3.8 shows the default signal-processing settings for the Freeverb left stereo channel. Processing for the right channel is obtained by adding an integer to each of the twelve delay-line lengths. This integer is called **stereospread**, and its default value is 23. (See the file **tuning.h** for all constants and default values used by Freeverb.) Different software distributions may include slightly different default values in **tuning.h**. The values in Fig.3.8 were found in the **ladspa-cmt-plugins** package (``Computer Music Toolkit") which is included in the Planet CCRMA distribution, and which is based on the June 2000 version of Freeverb, as of this writing. There are at least six more instances of Freeverb in the Planet CCRMA distribution alone.<sup>4.10</sup>

### Subsections

- Freeverb Main Loop
- Lowpass-Feedback Comb Filter
- Freeverb Allpass Approximation
- Conclusions

