Quiz #2

Quiz #2 on Friday. Covers text Ch. 1 - 4 and basic digital filtering.

Digital Synthesis of a Sound Field

In most recordings, we loose all directional information. Therefore we wish to somehow create and add in the experience of being in the concert hall. Since the characteristic of the hall can be divided into the direct sound, early reflections, and reverberation, we'll examine each individually to see how we will be able to digitally synthesize these.

Digital Synthesis of Early Reflections

The early reflections are simply delayed and attenuated versions of the direct sound. Since there are only a few early reflections after the direct sound, we have the possibility of using something similar to an FIR filter. A tapped delay line (TDL) is a special type of FIR filter in that it assumes most of the FIR coefficients are zero. In this case, we view the TDL as in the figure below where g_k are the tap gains and τ_k are the tap spacings (in samples) from the current sample. We assume $\tau_0 = 0$.

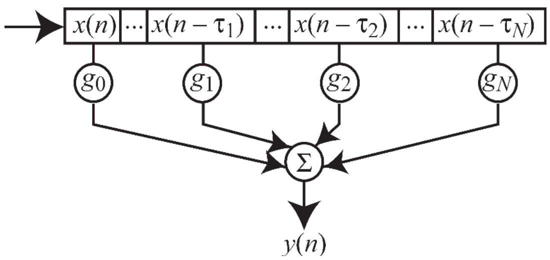

Figure: TDL

The output of the TDL is given by

$$y(n)=\sum_{k=0}^{N} x(n-t_k)\,g_k$$

Tapped delay line parameters for our system are give in Table 5.1 of the text.

Digital Synthesis of Reverberation

The reverberation in a full-size concert hall is usually in the range of 1 to 3 seconds. In principle, one extend the TDL put to this length with appropriate gains to create reverberation. However, there are usually hundreds or even thousands of reflections during reverberation time.

Example 1: Storage for two seconds of signal at fs = 48 kHz would require 96K of memory for each channel. The DSP5630xEVM has a total of 32K external memory. Internal memory varies according to the particular DSP56300 family member. For example, the DSP56302 has 7K+7K on-chip X- and Y-data memories.

Even if we had the available memory, computational complexity of this approach is prohibitive.

Example 2: If we assume 96,000 reflections (tap gains) at f_s = 48kHz, we require a minimum of 96,000 MACs per channel. Remember the upper computational bound was 1,250 instructions within a sample period @ f_s = 48kHz. Thus the computation required for the FIR approach to reverberation is unreasonable.

Comb Filter

(Straight from notes in text p. 169--173)

All Pass Filters

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(Straight from notes in text p. 174)

Implementation

(Straight from notes in text p. 174--176)

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