Master MVA
TP on sound effects and artificial reverberation

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1 Phasing

This first part is dedicated to the effect named phasing. This effect can be realised with a time-varying filter using a simple delay where the gain $a(n)$ varies (periodically) with time. The recurrent input-output equation is given by:

$$y(n) = x(n) + a(n).x(n-p)$$

where $y(n)$,$x(n)$ and $a(n)$ are respectively the output, the input and the gain of the system at time instant n (given in number of samples).

If we suppose a fix delay p (in number of samples) and a fix gain a, the transfer function of the full system is given by :

$$H(z) = 1 + a\ z^{-p}$$

1. Program the phasing effect with $a(n)$ following a sinusoidal variation of frequency f_a between the values $amin$ and $amax$. (One may choose $a(n) = B + A\sin(2\pi f_a n/F_s)$ with $B = (amax + amin)/2$, $A = (amax - amin)/2$ and where F_s is the sampling frequency).
2. Test the effect with different values of f_a, $amax$ and $amin$ (remember that values of a close to $a = 1$ give the strongest effect)

2 Flanger

This part is dedicated to the flanger effect. The implementation of this effect is quite similar to the implementation of phasing. Here, the time varying variable is the delay $p = p(n)$ and not the gain a as previously. This effect can then be realised with a simple delay line where the delay $p(n)$ varies periodically with time.

1. Write the filter recurrent equation

2. Program the phasing effect with $p(n)$ following a sinusoidal variation of frequency f_p between the values *pmin* and *pmax* . (One may choose $p(n) = B + A\sin(2\pi f_p n/F_s)$ with $B = (pmax + pmin)/2$, $A = (pmax - pmin)/2$ and where F_s is the sampling frequency).
3. Compute a value of p_{200} which induces the lowest zero in the spectrum of the fixed system ($a = 1$;$p = p_{200}$) to be at a frequency $f_{200} = 200$ Hz). Deduce appropriate values for *pmax* and *pmin*
4. Test the effect with different values of f_p, *pmax* and *pmin* (remember that values of a close to $a = 1$ give the strongest effect)

3 Artificial reverberation

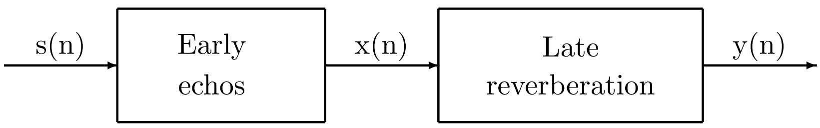

FIG. 1 – *Artificial reverberation scheme*

The aim of this part is to implement a complete artificial reverberation algorithm (see figure 1) by associating a specific module for early echoes (for reflexions up to second order) and a specific late reverberation module (using Schroeder reverberator).

3.1 Early echoes simulation

The simulation of early echoes will be done using the image sources method in a simple acoustic space, namely a cube of dimension [X,Y,Z]. The point O={0,0,0} corresponds to the point located at the bottom left of the space (see figure 2).

The source image method consists in determining the position of the images of the real source (there is one image per wall of the room, e.g. 6 image sources of first order).

The aim is then to realise an early echoes module with parameters:

- dimension of the cubic room ([X,Y,Z])
- position of source S
- position of microphone M
- sound speed c

which allows to obtain from a given source signal $s(n)$, a signal $x(n)$ with reverberation (only early echoes at this stage). For sound attenuation with distance, one may choose the following model:

$$\begin{aligned} a(r) &= \frac{1}{r} \qquad \text{pour} \quad r \geq 1 \\ &= 1 \qquad \text{pour} \quad r \geq 1 \end{aligned}$$

where r is the distance between an image source and the microphone M.

1. What can be done to improve the realism of the model for a real room?

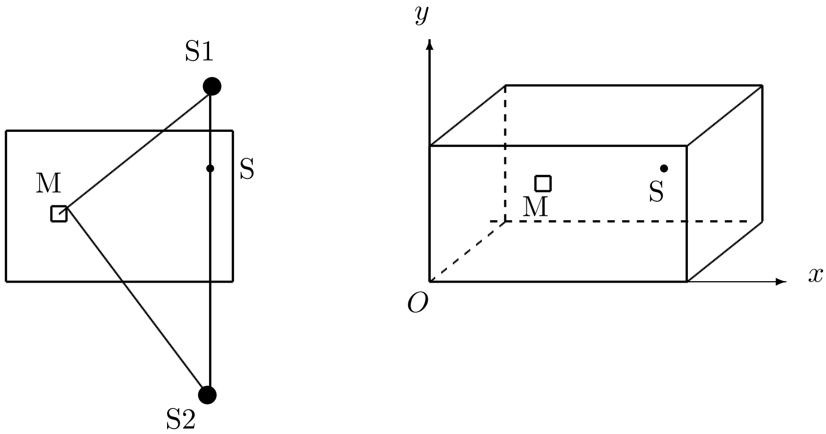

FIG. 2 – *Illustration of two image sources (first order echoes) in 2D and corresponding 3D representation of the simulated room.*

2. How many image sources would you have if you include second order echoes?
3. to your opinion, is it necessary to include higher order echoes? justify your answer.

3.2 Late reverberation: Schroeder reverberator

this part is dedicated to the implementation of Schroeder reverberator. The general scheme of this reverberator is given on figure 3.

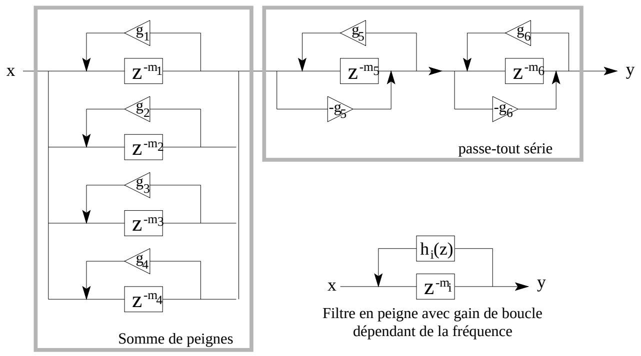

FIG. 3 – *Schroeder reverberant filter using comb filters and all-pass filters.*

For a comb filter, the reverberation time Tr is given by:

$$\frac{20\log_{10}(g_i)}{m_i T} = \frac{-60}{Tr} \tag{1}$$

where g_i is the feedback coefficient (or gain), m_i the delay duration (in number of sampling periods) of the cell i, and T is the sampling period. For an all-pass filter, the reverberation time Tr is given by:

$$T_r \approx 7T/\left[1 - g_i^{1/m_i}\right] \tag{2}$$

1. Knowing that it is better that each comb filter cell (resp. all-pass filters) has the same reverberation time, give the gain g_i which correspond to the following delays givne in

milliseconds (29,7 ms; 37,1 ms; 41,4 ms; 43,7 ms fro comb filters and 96,83 ms and 32,92 ms for all pass filters) as a function of the reverberation time.

2. Program and test your Schroeder reverberator on one of the given test signal for different room dimensions.
3. Test the reverberator with different parameters (keeping in mind that coprime integer values will lead to better results)