DESC 9115 Assignment 1: Lab Report 1 Delay Based Audio Effects (Resonator, Slapback, Echo)

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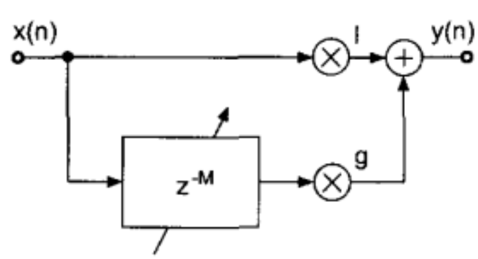
When a sound wave travels from the source it will disperse in different directions. The sound waves will continue until it is either out of energy or it hits a surface and reflects a different direction. When the receiver (an ear or microphone) hears the sound, it hears the direct sound coming straight from the sound source before hearing a delayed signal from a reflected surface. If the sound must travels further then there is more time for the direct signal to be heard, creating an echo, where as if it is close it will have more of a slap back or colouration effect. The parameters of the room are also important. If the sound is being reflected of many surfaces a flatter echo will occur. [1]

Some of the different effects that can be caused by delays are resonator, slap back and echo. One of the main differences between these effects is the delay time ($\boldsymbol{\tau}$), These effects are made by "comb filtering". Comb filtering adds a delayed signal of the input to itself causing what is known as interference where the waves will either sum or subtract. There are two main types of signal processor comb filters, The feed forward (FIR) and the feedback (IIR). [2]

Finite Impulse Response Comb Filter

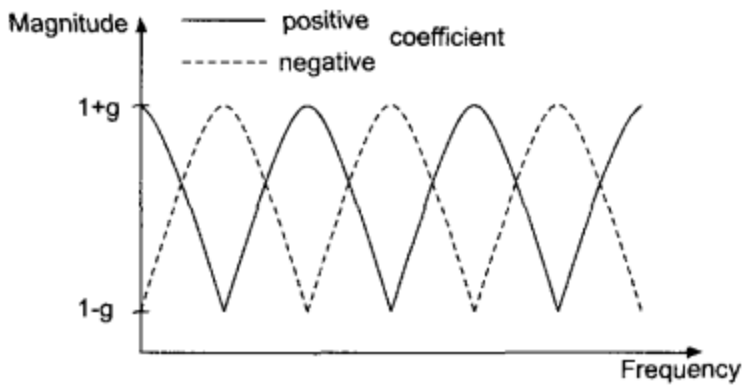
The FIR (Finite Impulse Response) simulates a delay of the input signal by a given time duration. The two variable parameters include the delay time and the relative amplitude of the delayed signal. The difference equation and transfer function are given by :

$y(n) = x(n) + gx(n-M)$
with M = $\boldsymbol{\tau}/f$
$H(z) = 1 + gz^{-M}$



FIR Comb Filter

The behavior of this type of comb filter is interesting in the way that the time domain affects the frequency domain. When the gain (g) has a positive value the filter amplifies all the frequencies that are multiples of 1/ τ (note that this is the formula to find the frequency of a wave where τ = period) and attenuates the frequencies in between them. The opposite happens when g is negative. Frequencies that are multiples of 1/τ are attenuated while frequencies between those are amplified. [2]



FIR comb filter Magnitude Response

When using an FIR comb filter there are two main effects that are oppositely proportional, one in the frequency domain and one in the time domain. When the delay signal is larger, we start to hear a repeated signal (e.g an echo) but as the time variable starts to get smaller and the signals start to blend and a definite repeat is no longer clear we start to hear the spectral effects of the comb filter in the frequency domain, causing a colouration of the sound. [3]

To create this basic FIR comb filter in Matlab the following code is used:

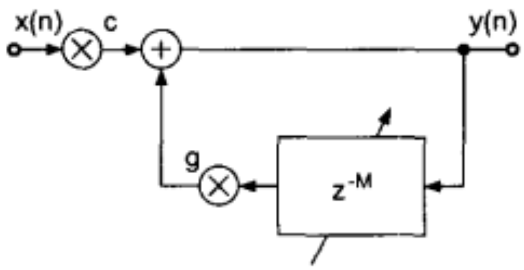
```
function y = fircomb(x, delay_in_ms, g, fs)


delay_in_seconds = delay_in_ms./1000;% Converts dealy time from ms to s

delay = round(delay_in_seconds.*fs);% Changes delay from samples to seconds

Delayline=zeros(delay,1);% memory allocation for length
for n=1:length(x);
    y(n)=x(n)+g*Delayline(delay);
    Delayline=[x(n);Delayline(1:delay-1)];
end;
```

Infinite Impulse Response Comb Filter

The IIR (Infinite Impulse Response) comb filter uses the feedback system producing and endless series of responses y(n) to the input signal. Each time the signal circulates through the delay line it is attenuated by the value of g. Because the system can cause a high amplification of the signal caused by the nature of the feedback system, it is sometimes important to scale the input signal by c. The difference equation and transfer function are given by:

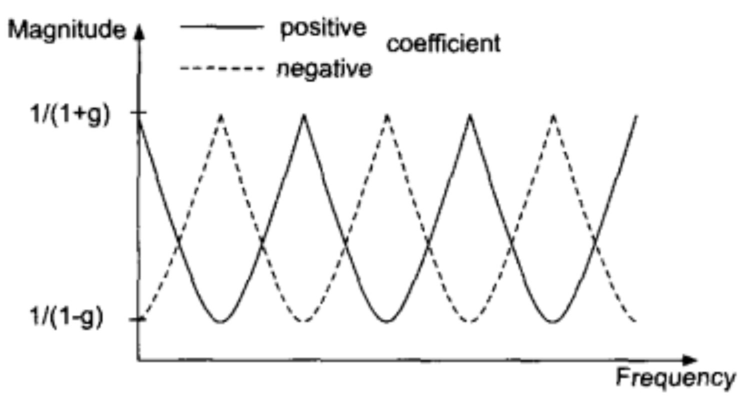
$y(n) = cx(n) + gy(n-M)$
with M = $\boldsymbol{\tau/f}$

$H(z) = c/(1 - gz^{-M})$



IIR Comb Filter

The time response in the IIR comb filter is infinite because instead of feeding a delayed signal forward, it feeds the input signal backwards continuously, this is called the feedback loop. Each time delay will come out with a gain (g) value numbered by the number of times the signal has gone through the loop. |g| must never equal 1 otherwise the signal would continually grow endlessly out of control. The IIR comb affects frequency in a similar way to the FIR comb, but the gain is varied between 1/(1 – g) and 1/(1 + g). As |g| approaches 1, the gain grows higher and the frequency peaks become narrower. [2] [4]



IIR Magnitude Response

To create this IIR comb filter in Matlab the following code is used:

```
function y = iircomb(x, delay_in_ms, g, fs)


delay_in_seconds = delay_in_ms./1000;% Converts delay time from ms to s

delay = round(delay_in_seconds.*fs);% Changes delay from samples to seconds

Delayline=zeros(delay,1); % memory allocation for length 10
for n=1:length(x);
    y(n)=x(n)+g*Delayline(delay);
    Delayline=[y(n);Delayline(1:delay-1)];
end;
```

Delay Based Audio Effects

By using these two comb filtering systems one can create different types of effects to a signal. If only one delay is needed (FIR) a variable on the delay time can be used to achieve our desired effect. If an IIR comb filter is used the feedback will enhance the effect and cause multiple delays. By using a modulation on the input signal the output sound can create new effects, but I have not done that here. [2]

Delay range (ms) (Typ.)	Modulation (Typ.)	Effect name
0 ... 20	-	Resonator
0 ... 15	Sinusoidal	Flanging
10 ... 25	Random	Chorus
25 ... 50	-	Slapback
> 50	-	Echo

Typical Delay Based Effects

Naturally the filter will add gain to the signal due to the system's structure, so the use of normalisation is important. In an FIR filter the most amplification that will occur is 6dB because it will only add the two signals together, however when using an IIR, there will be infinite amounts of waves adding and as |g| approaches 1 amplification grows. The L2 and L∞ norm are given by:

$$L_2 = 1/\sqrt{1 - g^2}$$

$$L_\infty = 1/(1 - |g|)$$

The normalisation coefficient $c = 1/L_\infty$ stops overloading the system, where as $c = 1/L_2$ keeps the broadband signals at approximately the same loudness. [2]

Conclusion

Digital delays caused by different types of comb filters can create a variety of different audio effects used mainly in audio processing and production, which can solve several problems including audio effects and delay compensation. By using different types of comb filter structures we can see how these effects are created and therefore have a better understanding of how they are used and manipulated to suit the user and how they deal with problems such as intrinsic amplification.

References:

[1] The Propagation of sound. 2012. The Propagation of sound. [ONLINE] Available at http://www.jhu.edu/virtlab/ray/acoustic.htm . [Accessed 01 April 2012].

[2] Udo Zolzer, 2002. DAFX:Digital Audio Effects. 1 Edition. Wiley, Chapter 3 Delays

[3] Feedforward Comb Filters. 2012. Feedforward Comb Filters. [ONLINE] Available at: https://ccrma.stanford.edu/~jos/waveguide/Feedforward_Comb_Filters.html . [Accessed 25 March 2012].

[4] Feedback Comb Filters. 2012. Feedback Comb Filters. [ONLINE] Available at: https://ccrma.stanford.edu/~jos/waveguide/Feedback_Comb_Filters.html . [Accessed 25 March 2012].