ASSIGNMENT 1: INITIAL TECHNOLOGY REVIEW
ENVELOPE CONTROLLED FILTERING

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ABSTRACT

An Envelope Controlled Filter (ECF) is a signal processing system which applies an input signal's average gain to a filter kernel, which in turn acts upon the input signal.

This review examines the algorithms that underlie ECF systems.

1. INTRODUCTION

Envelope Controlled Filters offer a highly interactive method of signal processing. This is due to the systems responsiveness to input signal variations over time, allowing the user to have a degree of control over the system's response on-the-fly. This is particularly useful when used in conjunction with musical performance.

1.1 Background

The ECF is commonly associated with music genres prevalent throughout the 1970s, most notably *funk*. The Mu-Tron III's liberal use on seminal recordings of the era established the ECF as mainstay system in signal processing.

1.2 Aim

The aim of this paper is to demonstrate understanding of ECF algorithms in a manner that is easily understood for the benefit of academic peers.

2. METHOD

In this section, we will comprehensively examine the way in which this ECF acts upon input *x(n)* to produce output *y(n)*. For reference, the systems tunable parameters (input arguments in MATLAB) are;

- *M* point window length (aka. *Q factor*)
- Cutoff frequency f_c
- Filter type (Low Pass, High Pass or Band Pass)
- Sampling frequency f_s (though typically defined by the input signal)

Note: These variables will be explained in detail during the *Filter Implementation* stage.

2.1 Envelope Following

2.1.1 Hilbert transform

MATLAB's `hilbert` function is used to output a gain value derived from the running average value of the input signal. It achieves this in a highly time-efficient manner by creating an envelope follower using the Fast Fourier transform and its Inverse.

`hilbert` takes the input signal and outputs a *ReX* analytic signal and the *ImX* Hilbert transform. The absolute value of the analytic signal is calculated and normalised, creating the envelope which describes the input's average positive amplitude.

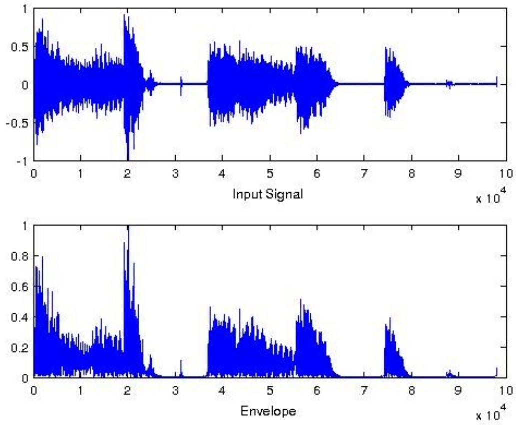
$$Hz(t) = H\,f(t) + i\,\hat{\hat{f}}(t) = \hat{\hat{f}}(t) - if(t) = -iz(t)$$

Hilbert transform of an analytic signal **(2.1)** [1]

where

Hz(t)	=	Hilbert transform of analytic signal
Hf(t)	=	Hilbert transform of input signal
if(t)	=	Imaginary N/2 signal
f(t)	=	Input signal
iz(t)	=	analytic signal

The IFFT moves this information back to the time domain where it may be convolved as the transfer function of the FIR filter, defining its gain parameter.



An input signal and it's envelope **[Fig. 2.1]**

2.1.2 Delay Samples

To allow for the envelope processing time, the input signal needs to be delayed by M/2 samples (ie. warm up).

2.2 FIR Filter Design

2.2.1 Finite Impulse Response

The ECF is executed as a Finite Impulse Response filter, acting upon the impulse for a fixed (*finite*) period, as opposed to an IIR which responds *infinitely* [2].

2.2.2 Ideal Low Pass Filter

The metaphorical *canvas* of the filter design is an i*deal* low pass filter. The reason for its idyllic label is due to all frequencies below the cutoff frequency being passed at unity gain, and all frequencies above are attenuated to 0. As a curve, this translates as the sinc function, given by Eq. (2.2) [3].

$$h[i] = \frac{\sin(2\pi f_C i)}{i\pi}$$

(2.2) [3]

The sinc function is shifted so that all values lie between 0 and *M.* This becomes the standard window length for the kernel.

2.2.3 Windowed-Sinc Function

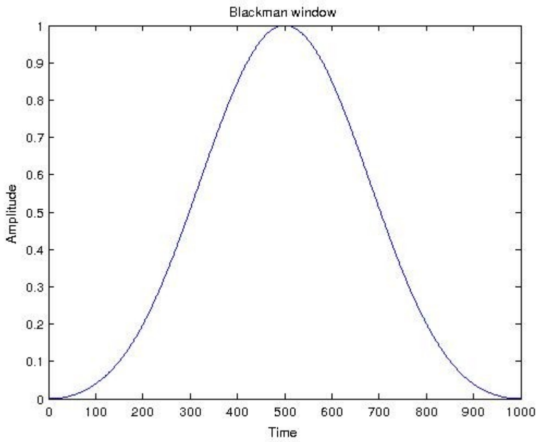
A Blackman window is imposed on the established low pass to smooth its transition between stopband and passband. This creates a transition band (*BW*), defined by *M*. This value in turn defines the size of the window, and hence the amount of samples processed per period through the whole system.

Considering the predefined parameters outlined earlier, we now have all of the information required to define the filter. The only information the kernel requires is the gain from the envelope follower.

While there are many different types of window functions to choose from, the Blackman window is superior for use within a windowed-sinc function. In comparison to a comparable window, the Hamming, the Blackman has a slow roll off, but excellent stopband attenuation [3].

$$w[i] = 0.42 - 0.5\cos(2\pi i/M) + 0.08\cos(4\pi i/M)$$

The Blackman Window function **(2.3)** [3]



[Fig. 2.2]

2.5.3 State Variable Filtering

State variable filtering utilises the corresponding relationship between lowpass, bandpass and highpass filters (2.4) to process the signal according to user selection.

$$y_l(n) = F_1 y_b(n) + y_l(n-1)$$
$$y_b(n) = F_1 y_h + y_b(n-1)$$
$$y_h(n) = x(n) - y_l(n-1) - Q_1 y_b(n-1)$$

(2.4) [2]

where

$x(n)$	=	input signal
$y_l(n)$	=	low pass filter
$y_b(n)$	=	bandpass filter
$y_h(n)$	=	high pass filter
F_1	=	f_c related tuning parameter
Q_1	=	resonance related tuning parameter

2.3 Tuning Parameters

2.3.1 Q-factor

M, more commonly known as the *Q* setting, is arguably the most significant variable of the system. The value of *M* defines the number of points in the roll off curve. This in turn determines the window size in relation to the sampling frequency through it's inversely proportional relationship to the transition bandwidth (*BW*). This relationship is approximated using the calculation in (2.5).

$$M \approx \left(\frac{4}{BW}\right)$$

(2.5) [3]

For example, if the sampling frequency is 48 kHz, and the value of M is 40;

BW = $0.1 f_s$
= 4.8 kHz

therefore

slower roll off = smaller M value
= larger BW value

Most people with a knowledge of DSP would be aware of the Q's affect on the resonance/ damping contrast of the filter. While this is a central aspect of the filter, if we look at the flow on effect of its value we can appreciate that that the Q/ M value is the heart of the ECF. By controlling the roll off severity, it is also defining how many samples are processed at a time.

2.3.2 Cutoff frequency

This leads us to the cutoff frequency. Like BW, this Hz frequency is expressed as a fraction of the sampling frequency. Being that the window is 0.5 f_s, the cutoff frequency value must lie between 0 and 0.5.

A good starting point is the center frequency;

f_c = $0.25 f_s$

therefore, if

f_s = 48 kHz

then

f_c = 12 kHz

2.3.3 LP/ HP/ BP

The filter selection simply chooses which equation from (2.4) to impose on the sinc function, with one exception. For the purposes of this system, a LP selection would not require any further information as it is the default selection of the ideal sinc function.

2.4 Filter Implementation

The filter kernel is the impulse response containing all of the required parameters for each M point sample period. Once the kernel is convolved with the input signal, the output signal is produced and, providing there is input at the next sample point, the whole process starts again... and again... and again... until the input signal discontinues.

$$h[i] = K \frac{\sin(2\pi f_c (i - M/2))}{i - M/2} \left[0.42 - 0.5\cos\left(\frac{2\pi i}{M}\right) + 0.08\cos\left(\frac{4\pi i}{M}\right)\right]$$

Windowed-Sinc kernel **(2.6)**[3]

3. CONCLUSION

The ECF is an impressive system to say the least. It's implementation of the Hilbert transform makes for a very computationally efficient method of DSP.

4. AKNOWLEDGEMENTS

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5. REFERENCES

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