DESC9115
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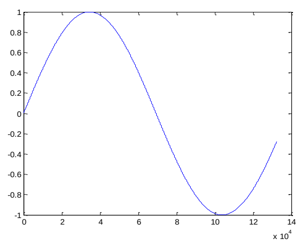
Digital Audio Systems Final Review

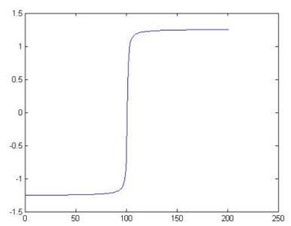
As lab report 2 was done on tape delay this review will go into more detail on simulating the characteristics of the tape itself through digital audio signal processing. The tape simulation processes performed in the tape delay function were simple and generic, simulating the effects of tape on a signal in a roundabout way. These processes performed were saturation and filtering. To come up with a solution for simulating the characteristics of tape a few more implementations of the previously stated processes will be looked at and also further implementation of digital processes.

Before looking at these processes sonic characteristics of tape should be identified. With tape recordings loud high-frequency transients in most cases do not survive as the tape is played subsequently over and over, hence the gradual degradation of transient detail as the tape is looped with a tape delay.(soundonsound) Another characteristic of tape is symmetrical soft saturation due to HF (high frequency) biasing (not always the case as will be shown in Abel, Lee's paper). The saturation is dependent on the signal level, non-linear distortion ceases at low levels (Klingelnberb, 1990). To look at this simply the level of the signal on tape will either have a linear or non-linear characteristic. Linear being in the lower levels and nonlinearities like soft compression (saturation) with higher input levels on to tape. With this in mind an improved (to that of lab report 2) saturation process can be performed by exploring other porcesses.

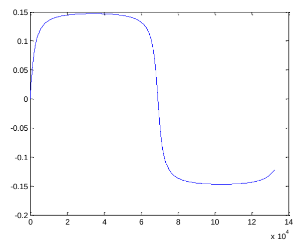
As in lab report 2 an input signal can be implemented into a weighted curve as a gain factor which is used to compress the signal, and so creating a saturated effect. In this particular case an inverse trigonometric function was used, arctan. (Mathematically illustrated in Lab report 2)



Input signal



Arctan function



Saturated signal

An improvement on this design would be one where the nonlinear function is only applied above a certain threshold to simulate the above characteristic of tape. A simple example of this is used to create an overdrive (symmetrical soft clipping) in the DAFX book (chapter 5) which could be implemented for tape saturation. The approach referenced by Zolzer [sch80] will be explained:

$$F(x) = 2x for 0 \leq x \leq 1/3$$

$$F(x) = (3 - (2 - 3x)^2)/3 for 1/3 \leq x \leq 2/3$$

$$F(x) = 1 for 2/3 \leq x \leq 1$$

Up to threshold of 1/3 the signal is multiplied by 2, this being in the linear region. The threshold between 1/3 and 2/3 the non-linear function is applied to the signal and soft compression is applied. If the input value is above the threshold of 2/3 then the value is set to 1. (Zolzer, Chapter 5, 2002). This method could be altered, with thresholds changed and the nonlinear function altered so as to come closer to measured saturation responses of tape.

The best way of getting as close as possible to the real thing is by convolution. A couple of papers will be drawn upon so as to gain a detailed insight into how this could be performed. Firstly a look at convolution. Any linear time invariant system may be represented in the time domain by its response to a specific signal called impulse. A linear time invariant system can be implemented as a digital filter by convolving an input signal with a system's impulse response (Smith, chapter 7, 1997):

$$y(n) = h(n) * x(n)$$

Where: y[n] is the output signal, x[n] is the input signal, h[n] is the system's impulse response.

Knowing this, it is easy to see how this could be implemented for simulation of tape. An impulse could be run through a tape recording system and its impulse used to accurately model the characteristics of that particular tape machine digitally. This is almost true. In many cases it is possible to create a sufficiently accurate simulation but factors such as the accuracy to which the impulse response could be determined, given the presence of noise in the signal process under analysis, need to be realised. (Kemp, 1999) In the case of the paper by Kemp which looks at simulation of non-linear audio processes, a step impulse signal is used to determine the impulse response, as it is said that it has significantly more energy in the lower frequency than the unit impulse. Another reason for using a step impulse is due to the fact that the system concerned will have different responses at different input amplitudes, especially in tape, with its linear nature at lower levels and nonlinearities at higher input levels (soft compression/saturation).Kemp looks at an altered convolution equation to be implemented. The reason for alteration is due to the non-linearity of the system being analysed. The above equation works for linear systems. The alterations involved to the final equation incorporate a selector function (selects particular impulse response for level input) and an interpolation function (to rectify a discontinuous switch between sample one and two). To illustrate this equation implemented, an evolution will be shown of how the two alterations are implemented:

Selector function:

$$S(x(n)) = 1 + x(n)/(fs/M)$$

Selector function incorporated into convolution:

$$y(n) = \sum_{k=0}^{L-1} x(n-k) h_{S(x(n-k))}(k)$$

The reason for the next step being the addition of an interpolation function is due to the fact 'where M<fs it can be seen that as input sample magnitude (x(n)) increases steadily from zero, response h(1), is used first then when the sample value reaches fs/M we have a discontinuous switch to response h(2).'(Kemp, 1999)

A linear interpolation function is used for the above reason.

$$p(x(n)) = (x(n))modulo(fs/M)/(fs/M)$$

The final equation for the non-linear convolution operation with both alterations incorporated:

$$y(n) = \sum_{k=0}^{L-1} x(n-k).(p(x(n)).h_{(S(x(n-k))-1)}(k)+(1-p(x(n))).h_{(S(x(n-k))-1)}(k)$$

Where: x:input signal, y: convolved signal, h:impulse response, S: selector function, p: interpolation function, L: length. (Kemp, 1999)

Another approach using convolution by Lee and Abel is used whose model is based on simulating the signal chain of the Echoplex EP-4 tape delay. Their design to simulate nonlinearities is called the LNLNL cascade, this is best described through an illustration.

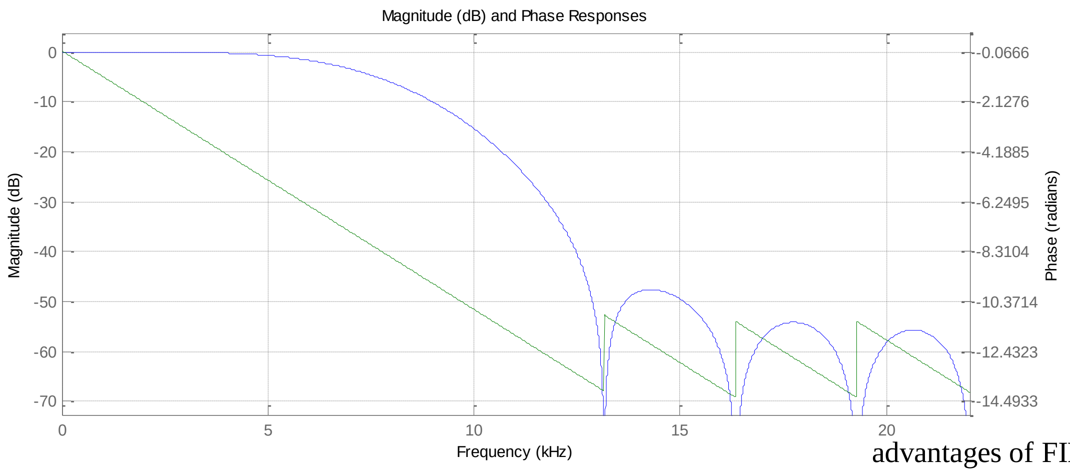
Input: S(t) ⟶ H1(t) ⟶ v(t) ⟶ H2(t) ⟶ u(t) ⟶ H3(t) ⟶ Output: r(t)

L N L N L

Where: L: Linear filter, N: saturating nonlinearity.

The reason for the structure of this LNLNL cascade is so that there is separate control over positive and negative going distortion. This is done by choosing the memory less nonlinearities to be one-sided saturators. V(t) saturates only large positive going inputs and u(t) saturates only large negative excursions of its input. (Lee, Abel, 2012). This control over the negative and positive distortion can readily reproduce asymmetrically saturated nonlinearities that may occur, and in this case with the Echoplex EP-4 did occur.

Now that a couple of cases of convolving the input signal to simulate nonlinearities (saturation) has been briefly looked at the next process which was involved in lab report 2 was the filtering of the signal. As was already explained tape will have a subtle effect on the frequency response of the signal, especially in the case of a tape delay there will be a narrowing in the range of the spectrum (depending on how much the signal is saturated where harmonics are created). In lab report 2 only one of the inbuilt FIR (finite impulse response) MATLAB low pass/high pass filters were looked at.

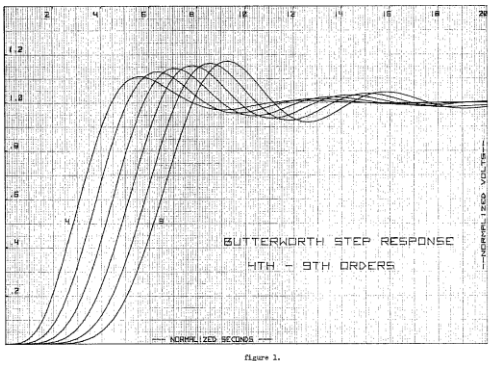


FIR filter used in MATLAB for Lab report 2

The advantages of FIR filters are that they off … disadvantage being that they typically require … ilar results. Only with the use of an FFT (fast fourier transform) are they competitive computationally.(Zolzer, chapter 2, 2002)

We will now look at a few more filter designs and simulations of analog filters which may get closer to the simulation of tape delay. The first case to be looked at is on a paper by Massberg. Here a digital Low-pass filter is designed based of an analog model using a bilinear transform. A bilinear transform is the transformation of a continuous time system to a discrete, or, simply analog to digital. Massberg finds a problem in using the bilinear transform in that when the cutoff frequency of the low pass filter reaches the nyquist limits severe warping of the response of the filter is introduced. To rectify this and to ultimately create a lowpass filter that is closer in magnitude response to that of the desired analog lowpass filter a pre warped first and second order low pass filter is applied to the bilinear transform.(Massberg, 2011)

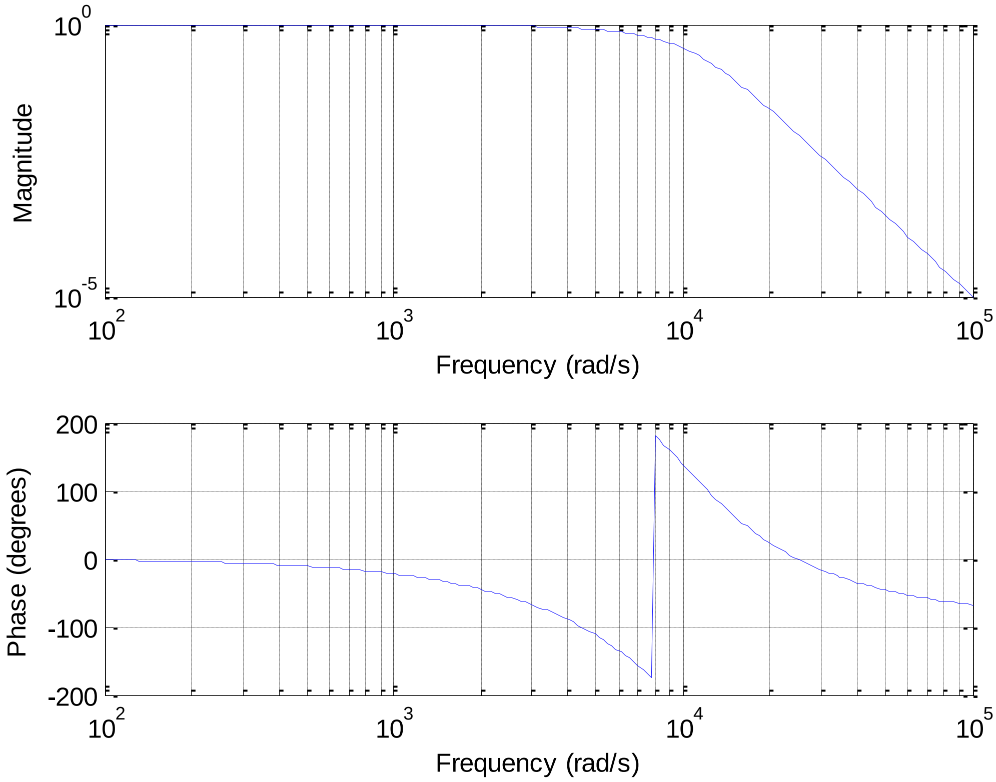
IIR (infinite impulse response) Filters such as the butterworth are well known for their maximally flat pass bands and good roll-off characteristics. But in Powell's paper he reveals that the butterworth filters are plagued by overshoots of as much as 17% for 9^{th} order filters. (Powell, 1981). These overshoots are best shown illustrated in a step response diagram:



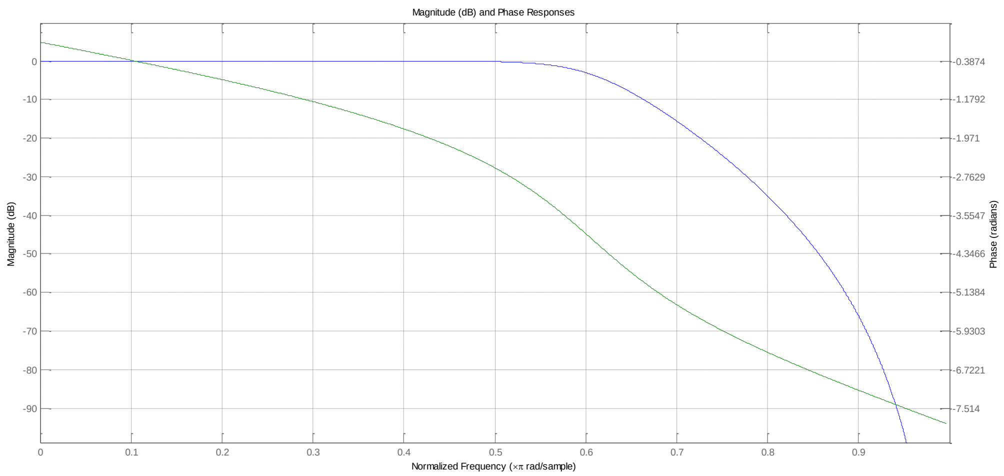
Step response diagram of Butterworth filter 4^{th}-9^{th} order from Lee Powell's paper 1981.

As can be seen in the diagram, as the order of filter increase so to does the overshoot, with respect to the butterworth filter. These slight ripples in the pass band will colour the input signal. Another filter which is refered to as one which you can resort to where overshoots wish to be avoided is the Bessel filter. These filters are designed to maintain an approximatley constant time delay for all frequencies in the pass band, providing overshoots of less than 1% (Powell, 1981). However it is later revealed that these Bessel filters encounter a sever problem at higher orders as their roll off is very shallow and as the function of a higher order filter is to have a sharper roll off immediatley after the corder frequency (-3dB) the use of a high order Bessel filter seems questionable. (Powell, 1981) In agreement with Powell: 'Lowpass Bessel filters have a monotonically decreasing magnitude response, as do lowpass Butterworth filters. Compared to the butterworth, Chebyshev, and elliptic filters, the Bessel filter has the slowest roll off and requires the highest order to meet an attenuation specification. (Mathwork)

Bessel Filter Magnitude and
phase response curves (5^{th} order)



Magnitude and phase of Butterworth filter 5^{th} order.



These two plots created in MATLAB illustrate the differences in characteristics between the Butterworth and the Bessel low pass filters.

Because tape has a subtle effect on the high frequencies as the signal is looped through the delay, a high order filter would not be suitable. Having a low order smooth pass band like the butterworth and even more so like the bessel low pass filter makes more sense in the implementation of a filter in the simulation of a tape delay.

This review looked at alternative processing stratedgies for simulating the filtering and saturation created by tape delay. From this a better understanding of how to implement these to simulate analog properties (tape delay and indeed other analog gear) was reached, from convolution, FIR/IIR filters and non-linear functions.

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