# DESC9115 2014 WRITTEN REVIEW 2

## Digital Audio Systems

## Lecture: William L. Martens

# Digital Flangers

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Abstract
This paper will look at the implementation of flanger in DSP. Describe what flanger is, how it works and methods of implementation. It was also look at the reasons why delay line interpolation is needed in flange systems and different methods of implementing it into the system. The paper will finally look at a method to circumvent the use of interpolation altogether through the use of all pass filters.

Delay line based or time based effects are all very closely related to each other. Different well known effect can be achieved by making small changes to signal routing, modulation wave type or even modulation rate. The most basic delay line based effect is vibrato, which is created by a moving a signal delay back and forth before the output. If you were to take a tap of the input signal before the delay and add it back to delayed signal before the output a flange effect is achieved. If you were to route the dry signal to left channel of a stereo system and the delayed signal to the right channel a basic chorus effect as achieved. Because all these effects are so similar there are industry standard delay offsets and depth limitations that define apart. Vibrato has no delay offset and is between 0-3ms in depth being modulated be a sub sonic low frequency sine wave. Flange also has no delay offset and have a depth of 0-2ms (Zolzer, 2002). However not every effect designer follows these standards and sets parameters based on personal taste.

Flange is a time based effect that functions by modulating the delay signal that is added back to the original signal with an arbitrary gain applied to it. By adding two identical signals together with a small amount of delay creates a series of peaks and dips in the frequency response. Where these peaks and dips occur in the frequency domain depend on the amount of delay($t$) in the time domain and the gain($g$) factor. For positive $g$ values the peaks occur at frequencies that are multiples of $1/t$ and dips at frequencies that lie in between. (Zolzer, 2002). This effect is known as comb filtering and is the basis of flanging. If the delay time is varied by a continuous low frequency sinusoidal signal, the comb filter moves up and down the frequency spectrum which creates the effect of flanging.

Analogue flanging is accomplished in many ways. In the days of tape, engineers would achieve flanging by recording the same material onto two tape reels and summing the playback. They would then press their thumb against the flange of the second tape reel causing the tape to slow down with respects to the first tape reel. The audible result is a moving comb filter or flanger. The effect driving it name from the method used to produce it. Flange was very popular in psychedelic music of the late sixties and early seventies. The effect was made famous by band like the Beatles and Jimmi Hendrix.

Flange is implemented in the digital domain by first creating a simple FIR comb filter. To create the moving comb filter the delay is modulated by a sub audio signal (below 20Hz).

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Fig.1

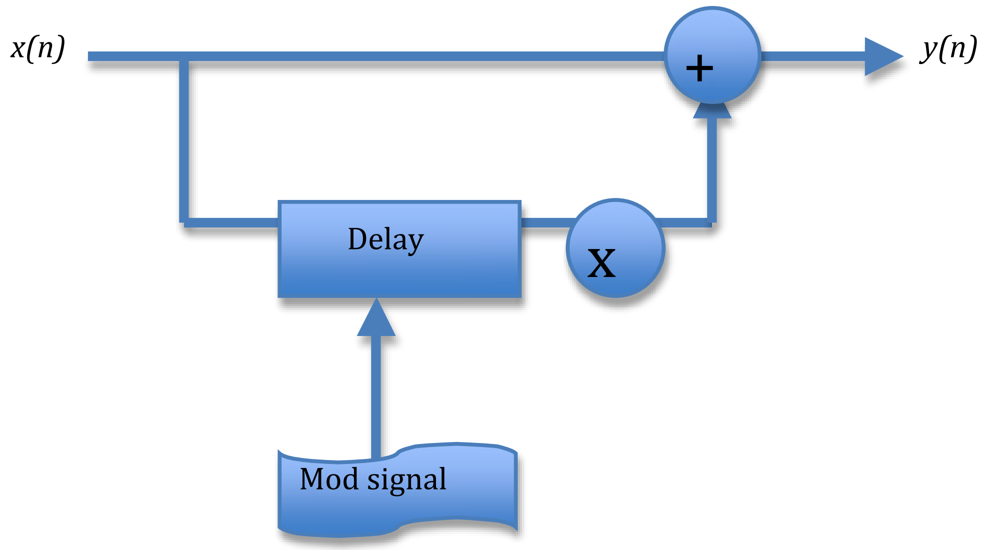



Fig.1 illustrates a block diagram of a mono flange system as a FIR filter. A split of the input signal, *x(n),* is delayed then multiplied by a gain factor to control the mix of the filter and is summed back with the original signal at the output *y(n)*. This can be expressed by the equation

$$y(n)= x(n)+x(n-d(n))g$$

Where *d(n)* is the delay length in samples.
And *g* is the gain value

The amount of delay is varied by the mod signal. This is a low frequency that moves the delay line back and forwards in time to create the moving comb filter effect. The mod signal has three main parameters: the wave type, the frequency and the depth. The wave type defines the shape of the wave either sine, triangle or sawtooth. The frequency controls the rate of the modulation wave and the depth is the amplitude of the modulation wave. This system is reflective of the old analogue method of creating flange with tape.

Flange can also be implemented using a recursive filter method such as IIR. However due to the recursive nature of the filter an extra gain stage *c* is needed to scale the input level (Zolzer, 2002). This system can be express by the equation

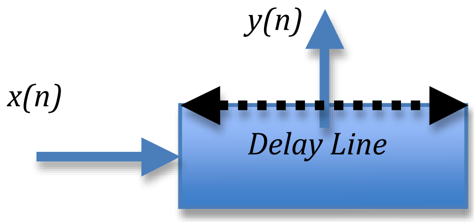
$$y(n)=cx(n)+gy(n-M) \quad \text{where } M=t/fs$$
$$H(z)= c/(1-gz^{-M})$$

Zolzer, 2002

This extra gain stage is added at the beginning of the system before the delay signal is added back to the input. It should also be noted that values of *g* should not exceed 1, otherwise the system would become unstable and the system gain would grow endlessly. While FIR filters are more accurate they are computationally more expensive which make IIR a better choice for real time application.

Employing a feed forward method of flange (such as fig.1) has a potential problem that only can exist in the digital domain (Smith, 1982). The problem is one that is due to the discrete nature of digital signals. If the LFO is to modulate the delay signal smoothly delay-line interpolation is needed to avoid signal discontinuity. As the delay signal is swept back and forth it passes through values of *n* that are non-intergers. Dattorro (1997) explains why delay line interpolation is needed with this diagram of a simple vibrato system:

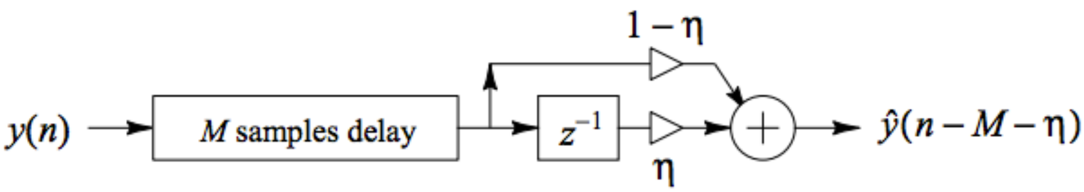
Fig.2




In this system the delay line is 2048 samples in length. At every sample time *n* a new sample is added from the input *x(n)* and the oldest sample is discarded. The output is rarely the last sample in the delay line, Dattorro has shown this by not putting the output to the right of the delay line block, as would be the convention, but has put it at an arbitrary position above the delay line block. This position is called the tap center, and because the arbitrary output position is smoothly moving about the tap center real time interpolation is needed to fill in the information in between the discrete samples.

To have achieve a good result the interpolation device must have flat frequency response, linear phase response and transient free response to variations of the delay length (Rocchesso, 1998). The simplest interpolators are linear delay line interpolators (fig.3) and allpass delay line interpolator (fig.4).

Fig.3




Smith, 2014

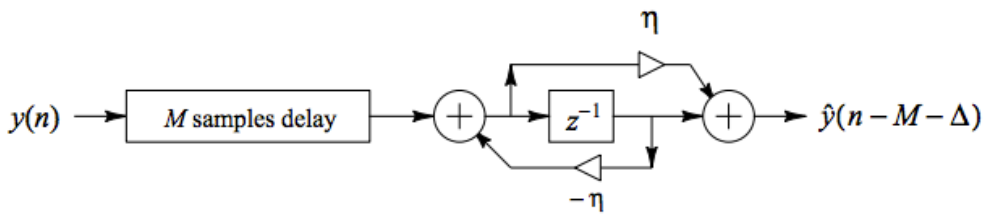
The linear interpolation system is a feed forward system that is expressed by:

$$\hat{y}(n - \eta) = (1 - \eta) \cdot y(n) + \eta \cdot y(n - 1)$$

Where η is the fractional delay.

While the computational cost of linear interpolation for performance ratio is very good it does have a few drawbacks: Amplitude distortion and modulation, phase distortions and aliasing (Dattorro, 1997). The amplitude distortions are describes as a muffled effect as frequency rise and the amplitude modulation are describes as an audible flutter that is specially noticeable in the high frequencies. The phase distortions are due to the fact that delay responses are not constant between each polyphase filter. The aliasing occurs when there are large pitch changes in the input signal. Using allpass filters can solve the drawback of amplitude distortion and modulation. Dattorro (1997) states that using allpass filter to perform interpolation is somewhat foreign but he justifies his decision with a comparison to linear filter and decides that they work much better and are computationally cheaper to run.

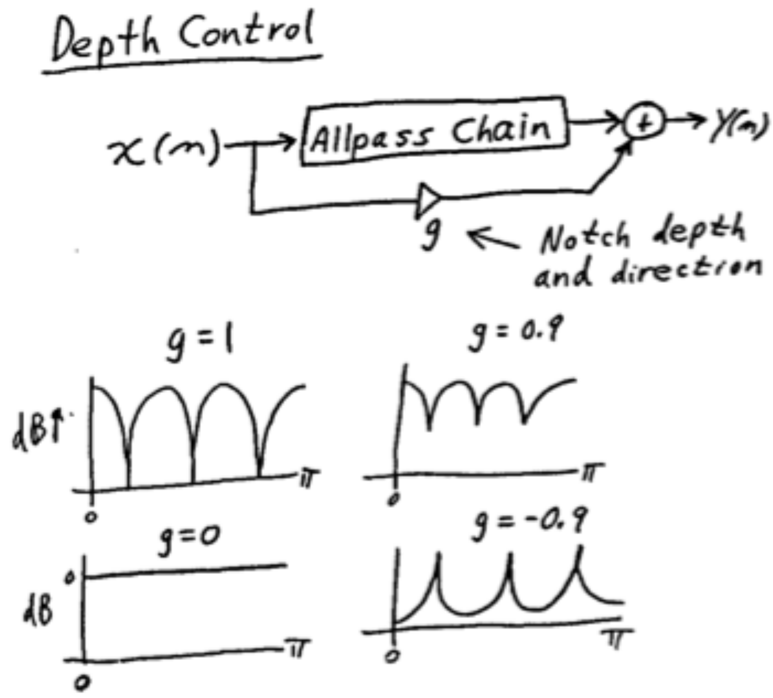
Fig.4




Smith, 2014

To circumvent the need for delay line interpolation Smith suggests using a series on allpass filters to create phaser. While this is not flange in the strict sense as it is introducing a phase difference and not a time difference it does remove the need for interpolation as the delay is a function of phase and not time. The system has notches wherever the phase of the allpass chain is pi. The depth of the notches are modified by changing the gain of the dry signal. Whether the gain is positive or negative dictates the direction of the notches (fig.5) The width of each notch is controlled by the damping factor of the corresponding allpass filter (Smith, 1982). Each allpass filter has unity gain making the system very stable like an FIR delay line system.

Fig.5




Smith, 1982

Flange can be implemented into digital systems many different ways. It can be as simple as creating a delay and add system where the delay line is modulated by a LFO. However this method can lead to issues of discontinuity in the final output signal. This is caused by the fact that the delay line is a factor of time *n* and needs delay line interpolation to fill in miss part of the signal when delay passes in between samples. Linear interpolation is a simple and easy method to rectify the discontinuities of the signal. However this does not perform so well at high frequencies and causes audible flutter. Dattorro (1997) suggested using allpass filter to perform interpolation and achieves better results in the high frequencies with similar computational power. Smith (1982) suggested using allpass filters in place of a delay line to create delays in the system. This method is phase dependent and not time dependent, theoretically circumventing the need for interpolation, as all movements will have a phase value. However this method is now phasing and not flanging. Phasing has a different timbral effect to flanging and Smith's proposed method would not satisfy an artists desire to add flange.

References:

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