Audio Effects Portfolio

Kyle Daruwalla & Jacques St. Louis

May 12, 2016

1 Chorus Effect

1.1 Description

This mono effect combines a track with one or more delayed copies. There is a separate gain control for the dry copy and each of the delayed copies. The delay block accepts fractional delays from a local oscillator (usually at a frequency of 0.1 Hz) where each delayed copy has an LO at a different phase. As a result, the listener will perceive multiple voices or instruments playing together (like a chorus).

1.2 Applications

The chorus effect is an easy way to add depth to a track. In particular, it is an artificial method to simulate double-tracking, a common technique used by artists to create a "fuller" sound.

1.3 Principles of Operation

The chorus effect is composed of several variable delay lines controlled by an LFO. Typically, the original, dry signal is added to the delayed, wet signals. The effect can come in two variants - standard chorus or multi-voice chorus. The standard chorus contains only a single delay line. The multi-voice chorus can contain several delay lines, where each delay line's LFO differs in phase. Figure 1 contains a block diagram for a multi-voice chorus effect with two delay lines.

1.4 Implementation

We implemented a standard chorus (chorus.m) and multi-voice chorus (chorus_multi.m) with four delay lines in MATLAB. The standard chorus had a single delay line with an LFO at 0.08 Hz and peak delay of 30 ms. The multi-voice chorus featured four LFOs that were all operating at 0.08 Hz and peak delay of 30 ms. However, the first LFO was in-phase with the original signal, the second was 45° out of phase, the third was 90° out of phase, and the fourth

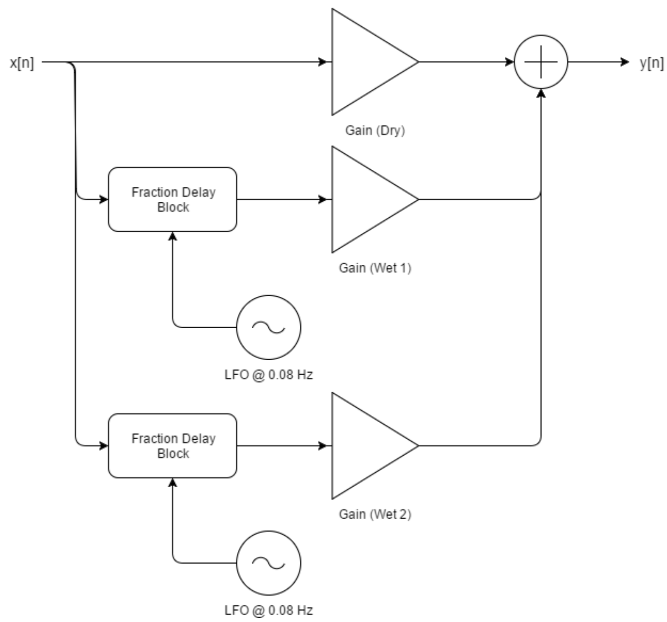

Figure 1: Block Diagram of Chorus Effect

was 135° out of phase. The first and second LFOs were hard panned to the left channel, and the third and fourth LFOs were hard panned to the right channel. The end result is a richer, fuller sound.

1.5 Demo and Discussion

The original audio is an electric guitar track that is mono and panned center. We found the best results were obtained for an LFO frequency of 0.08 Hz and a peak delay of 30 ms. These settings can be heard for both the chorus and the multi-voice chorus effects.

We also examined how the effect changes when parameters are varied. Since the LFO frequency is less than 0.1 Hz, changing the frequency is not as noticeable as an effect like flanger. However, there is a subtle change in the level of depth or richness added by the effect to the track. We created output tracks for LFO frequencies of 0.01 Hz, 0.02 Hz, 0.03 Hz, 0.04 Hz, 0.05 Hz, 0.06 Hz, 0.07 Hz, 0.08 Hz, 0.09 Hz, 0.1 Hz.

Varying the peak delay has a more pronounced effect than varying the LFO frequency. In particular, at a peak delay of 10 ms, the synthetic quality of the flanger effect can be heard. As we vary the peak delay up to 20 ms, the synthetic quality is faintly audible; it is complete gone at 30 ms.

1.6 Further Exploration

To see this effect used in a real world application, check out this YouTube video that uses a chorus pedal on a guitar.

2 Flange Effect

2.1 Description

The flange effect is very popular among musicians, as it adds interest to a track in a periodic way. Flanging sounds as if an original instrument's sound is changed from a natural to artificial and back.

2.2 Applications

The flange effect is often used to add a periodic, synthetic sound to a track. This adds an interesting sound variation to most any musical track.

2.3 Principles of Operation

The flanger is quite simple in concept: it consists of a variable delay which is controlled by a low frequency oscillator (LFO). The delay output is then summed with a non-delayed version and output to the destination.

2.4 Implementation Notes

To implement this effect, we used MATLAB (see Flange.m). Starting with the basic audio file implementation from class to handle importing and playing the audio files. We built upon this and added both a new line in the step function for adding the two signals to be produce together with varying gains. A block diagram can be seen in Figure 2.

2.5 Demo and Discussion

Flanging can be used in a variety of ways to produce a wide range of sound effects from the same track.We adjusted the maximum delay using a 4ms delay and an 8ms delay using a 0.4Hz LFO. We then tried used a 1Hz LFO with a 4ms and an 8ms delay.

The flanging sound is very noticeable, but not extremely deep. Many flangers have feedback in their delay line, creating resonances in the track which can increase the effect's presence in the track.

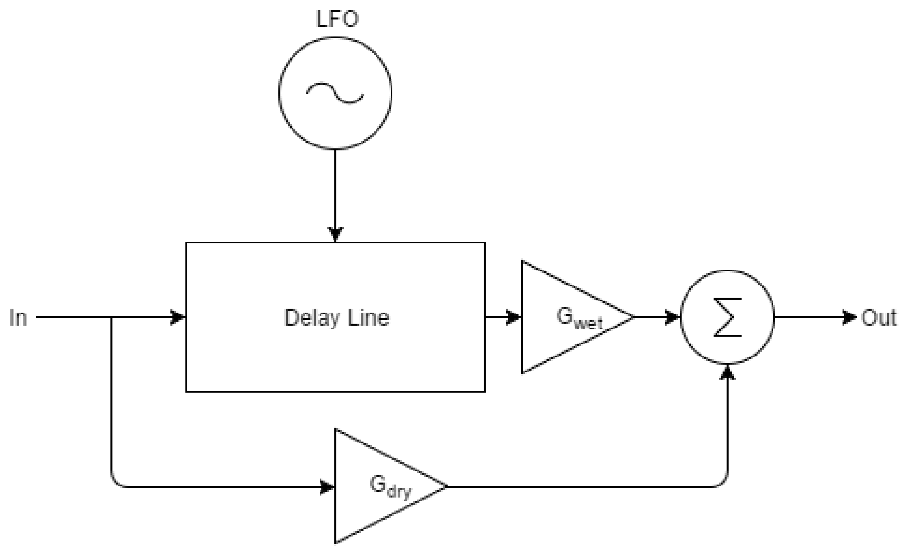

Figure 2: Flange Block Diagram

2.6 Further Exploration

Something to investigate is how this effect would change if it also had a chorus aspects, placing the flange effect at different rates on different chorus channels. This would be an interesting demonstration of two effects from this chapter. A cool way to further investigate this effect is to watch this neat YouTube video about the flange effect here.

3 Phaser

3.1 Description

This mono effect creates adds a futuristic sweeping sound to a sample track. This is done using a series of all-pass filters which maintain the magnitude of a signal while applying a nonlinear phase shift. In order to create a sweeping effect, the center frequency of each all-pass filter is varied with an LFO.

3.2 Applications

Typically, phasers are used by guitarists to create an electronic or unnatural sound for tracks that are meant to sound futuristic or ethereal.

3.3 Principles of Operation

A basic phaser is composed of a series of all-pass IIR filters. However, a single filter alone cannot create the notches in the output spectrum that are characteristic of phasers. In an analog implementation, four first order filters or two second order filters. Each all-pass filter maintains the magnitude of the signal, but it applies a nonlinear phase shift to the input phase. The overall output phase is the sum of the phase shift added by each filter. This filtered signal is then amplified and added to the original signal. This create multiple notches in the output spectrum as seen in Figure 3. Each notch represents destructive interference between the filtered signal and original signal. In order to move the

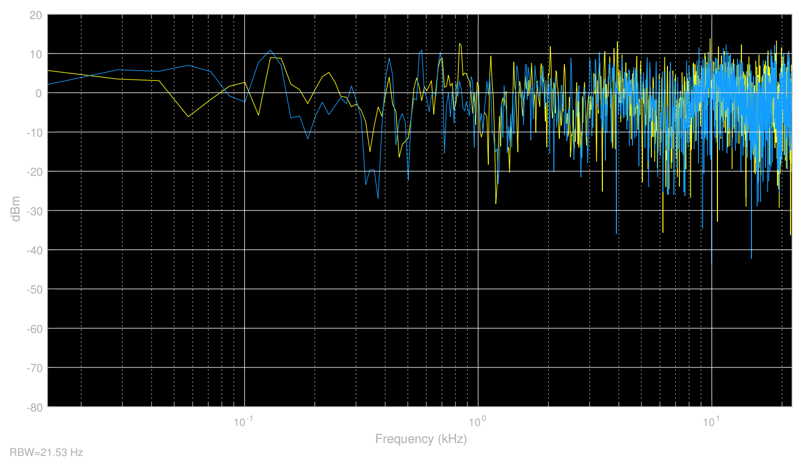

Figure 3: Output Spectrum of a Phaser Effect

notches, the center frequencies of each filter is varied using an LFO. Each filter has the same center frequency. The LFO output is given by Equation 1 and shown in Figure 4.

$$f_c[n] = f_{min} \left(\frac{f_{max}}{f_{min}} \right)^{triangle(\omega_{LFO} n)} \tag{1}$$

In order to improve the Q of the all-pass filters, a feedback path can be added. The full block diagram with feedback is shown in Figure 5.

3.4 Implementation

Our implementation for phaser is unique, because it uses the STFT to move the time domain input signal into the frequency domain, then apply the appropriate phase shift with simple addition, then move the filtered spectrum back into the time domain. Typically, a phaser implemented in MATLAB would make use of the biquad filter command instead of using an STFT and frequency domain phase processing.

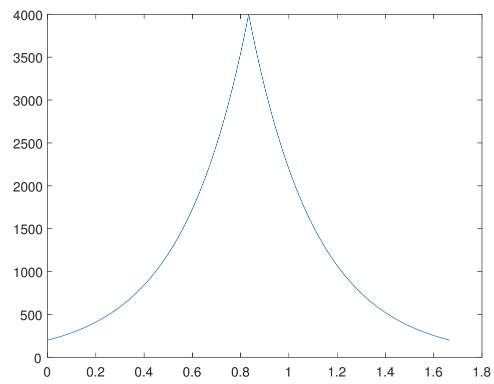

Figure 4: Phaser LFO Output Between 200 Hz and 4000 Hz

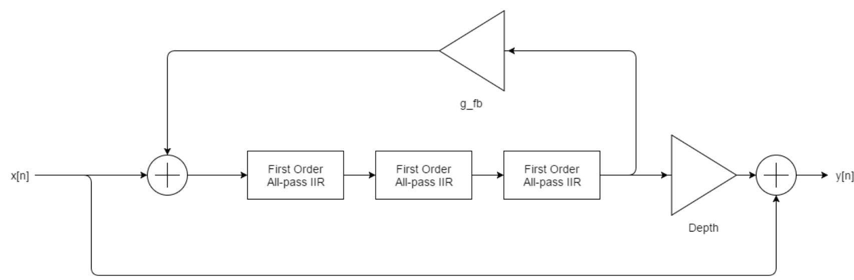

Figure 5: Phaser Block Diagram with Three Allpass Filters

3.5 Demo and Discussion

The effect is most easily heard in output for colored noise. However, this is not very interesting. Applying the effect to an original guitar sample results in a more pleasing output. If a not so subtle effect is desired, then the LFO frequency can be increased from 0.2 Hz to 0.8 Hz or even 1.2 Hz. The effect can also be applied with no feedback, though the differences are less subtle.

3.6 Further Exploration

You can check out an obvious use of phaser in Billy Joel's Just the Way You Are. A more subtle use of the effect can be heard in Led Zeppelin's Kashmir. It may not be clear at first, but if you listen to the drums, especially the crash of the symbols, you can hear a soft phaser being applied.

4 Tremolo

4.1 Description

This mono effect mimics tremolo or vibrato created by musicians on instruments like violins or guitars. Tremolo is create by exploiting the same technique for amplitude modulation (AM) with a low frequency carrier signal.

4.2 Applications

Tremolo can be used in mixing applications to create a natural vibrato to a flat sounding track. However, its use is extremely apparent in electric guitar tracks that use it with a higher rate.

4.3 Principles of Operation

The operation is fairly similar to amplitude modulation. However, while AM signals might have carrier frequencies in MHz, tremolo carrier signals only range between 1 and 10 Hz, typically. The carrier signal is defined in Equation 2. A block diagram on tremolo can be found in Figure 6.

$$m[n] = 1 + \alpha \cos(\omega n) \tag{2}$$

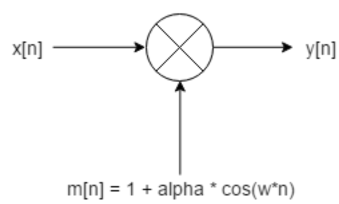

Figure 6: Tremolo Block Diagram

4.4 Implementation

Tremolo is implemented in MATLAB using an LFO to generate the $\alpha \cos(\omega n)$ term in Equation 2. Time domain processing is then applied to mix the carrier signal with the input signal.

4.5 Demo and Discussion

The effect was applied to this original guitar track. The effect can be heard for a carrier frequency of 5 Hz. To hear how changing the carrier frequency changes the output, the effect was applied for 1 Hz, 10 Hz, and 20 Hz as well.

4.6 Further Exploration

A great use of tremolo can be heard in Link Wray's Rumble. Though the effect is barely in use at first, as the song progresses, the guitarist turns up the depth (α) on the tremolo until it is extremely obvious.

5 Ring Modulation

5.1 Description

This mono effect modulates a track with carrier signal to create a wavering robotic sound.

5.2 Applications

Ring modulation can be used to added a robotic sound to a track. Often, it is used on speech or guitar tracks.

5.3 Principles of Operation

Ring modulation is done by modulating the input signal with a carrier signal whose frequency is in the range of the audio. The carrier signal is described by

$$m[n] = 1 - \alpha + \alpha\cos(2\pi f_c n)$$

where α is the depth of the carrier signal and f_c is the carrier frequency. Using $\alpha = 1$, suppressed carrier amplitude modulation is achieved. Decreasing α will add more of the DC term back into the carrier signal. The carrier frequency typically ranges between 10 Hz and 1 kHz. The output is then given by

$$y[n] = x[n]m[n]$$

A block diagram can be seen in Figure 7.

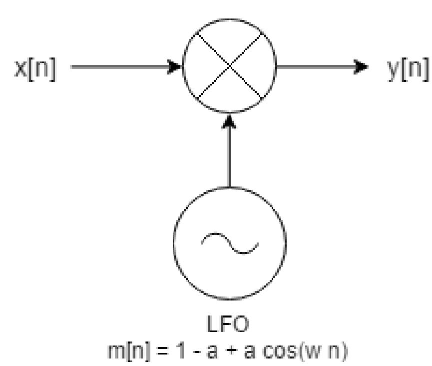

Figure 7: Block Diagram of Chorus Effect

5.4 Implementation

We used MATLAB to implement the ring modulation effect (see ring_modulation.m). An LFO object was used to generate the signal $\alpha \cos(2\pi f_c n)$. This was then used to create the signal $m[n]$. We multiplied the generated carrier signal with the input $x[n]$ to produce the output. Each channel of a stereo track was modulated individually.

5.5 Demo and Discussion

The effect is most obvious when comparing these original vocals to the modulated output (LFO at 250 Hz and $\alpha = 0.5$).

However, a more pleasing use of this effect is achieved when it is applied to this original bass track. To study the effect, the different values of α were tried: 0.1, 0.5, 0.8, 1.0.
Comparing the $\alpha = 0.1$ and $\alpha = 1.0$ tracks, the effect of the depth parameter can be heard. With a larger α, the rich spectrum of the audio is lost, and several notes sound flat.

Additionally, we can vary the frequency of the LFO. Changing the frequency will change the rate of the wavering in the output audio. At higher frequencies, the wavering is occurring so fast that the output audio sounds robotic. We attempt frequencies of 10 Hz, 50 Hz, 100 Hz, 1000 Hz.

5.6 Further Exploration

To learn more about this odd effect, check out this YouTube video.

6 Distortion (Clipping)

6.1 Description

This mono effect mimics purposefully distorts a signal to create nonlinear effects. Two types of distortion exist - soft clipping and hard clipping. Soft clipping mimics an older era of vacuum tubes, whereas hard clipping mimics the harsher sound of a transistor amplifier.

6.2 Applications

Distortion is common technique used in modern or rock music to create a harsher, richer sound.

6.3 Principles of Operation

Distortion is simply applying a nonlinear gain to a signal so that the input-output transfer curve is altered. Soft clipping has a smoother, continuous transfer curve, while hard clipping is discontinuous. Figure 8 shows the transfer curves for soft and hard clipping. Soft clipping can be implemented using

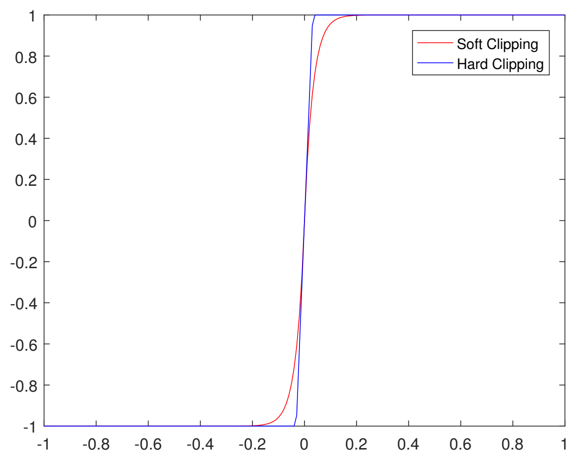

Figure 8: Input-Output Transfer Curves for Soft and Hard Clipping

Equation 3, and hard clipping uses Equation 4.

$$y = \operatorname{sgn}(x)(1 - e^{-|Gx|}) \tag{3}$$

$$y = \begin{cases} -1 & Gx \leq -1 \\ Gx & -1 < Gx < 1 \\ 1 & Gx \geq 1 \end{cases} \tag{4}$$

6.4 Implementation

Distortion was implemented in MATLAB using the equations in the previous section. The gain is specified in dB, and the script reinterprets this parameter as the linear gain, G.

6.5 Demo and Discussion

The effect is best heard on the “Hero” vocal track. Soft clipping was applied for 10 dB, 30 dB, 50 dB. The output sounds particularly harsh at 50 dB. However, it is possible to distinguish between an extremely loud soft clipping signal and the harshness of hard clipping, even at only 30 dB. Figures 9 and 10 show the output of vocal sample for both distortion styles at 30 dB.

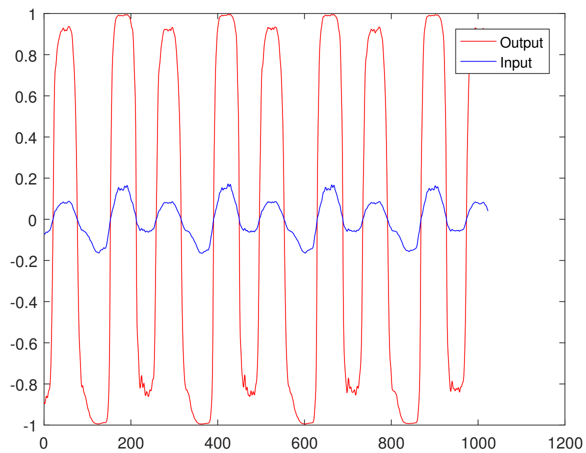

Figure 9: Output at 30 dB with Soft Clipping

6.6 Further Exploration

Though distortion can be found in almost any rock song, a great song to contrast a distorted guitar versus a clean guitar is Led Zeppelin's Over the Hills and Far Away. Around 1:40, you can hear some cleaner guitar, but very soon, Jimmy Page transitions to the harder riff of the song, and the distortion kicks in.

7 Frequency Bin Shifting

7.1 Description

Pitch Shifting with frequency bins is a way in which one can change the pitch of a given track. The "bin" application stems from taking the Fast Fourier Transform (FFT) of a given waveform. An advantage to this processing technique is its speed, though it must be noted the outcomes do not always sound the best compared to time stretching/compressing. Another plus to using frequency bins for pitch shifting is the ability to define frequencies grouped into each bin and the number of bins to shift by, allowing precise control of the shifting degree.

7.2 Applications

This process is often used in certain types of music and audio production. Some examples of pitch shifting can be found music, making certain vocalists voices sound deeper than they normally are. In film, pitch shifting may be used to make an actor sound younger than they are, such as making a voice higher to sound younger.

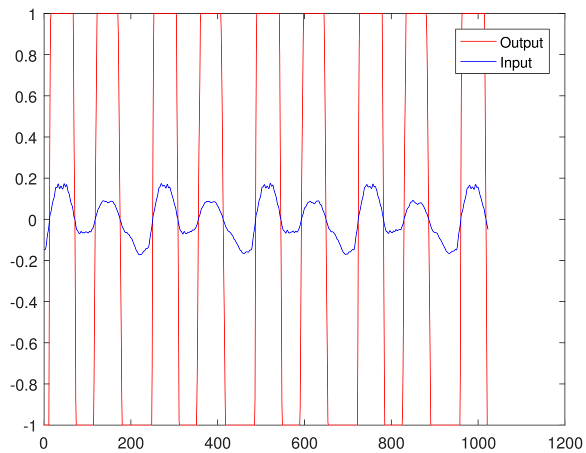

Figure 10: Output at 30 dB with Hard Clipping

7.3 Principles of Operation

THe basic idea here is that by taking the FFT, one can break the audio waveform into levels of specific frequencies.By taking the amplitudes and phases in these bins, changing the pitch is as simple as moving a bin from one location to another, high for a higher pitch, lower for a lower pitch. THe degree of shift determines how much of a change will take place, and the more numerous the bins over a given range of frequencies the better, giving the user greater control over the amount of shift as well as increased resolution in frequency representation and reconstruction.

7.4 Implementation Notes

diagram can be seen in Figure 11.

7.5 Demo and Discussion

In using frequency bin shifting, the shift factor has one of the most profound effects, other changes have significant changes as well. First we set the hop size to 20ms and the shift factor to change, first being 5 Bins and another at 10 Bins. We then changed the hop size to 50ms, then repeated with 5 Bins and again at 10 Bins.

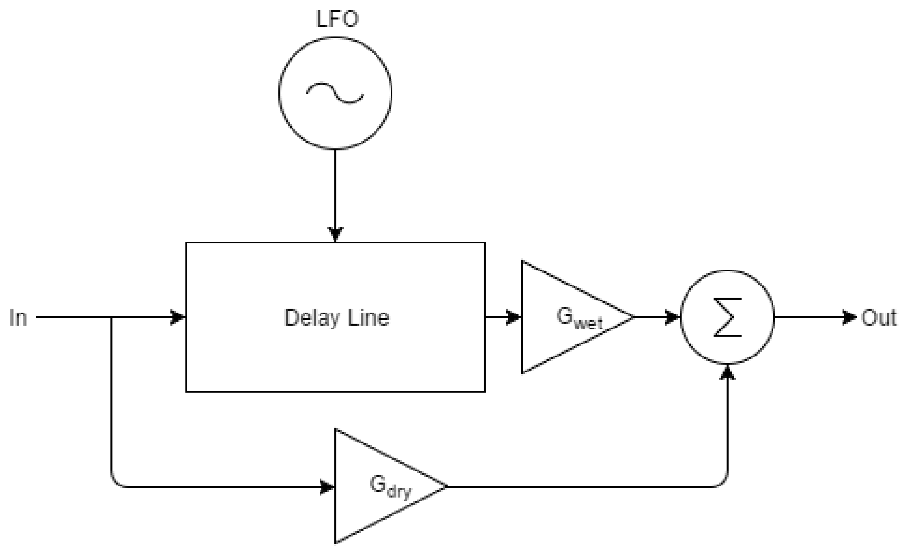

Figure 11: Flange Block Diagram

7.6 Further Exploration

Pitch shifting is especially active in the guitarist community. For further work and discovery in the FFT bin shift and more about the FFT, check out the link here. The web page provides good figures and explanations to understand just what each step in the program is accomplishing.