Delay Based FX Plugins in MATLAB

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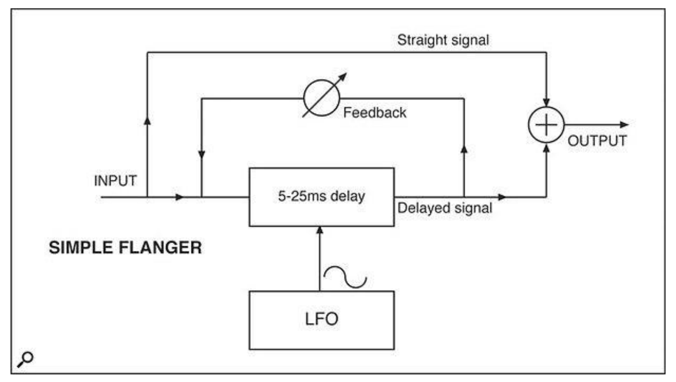
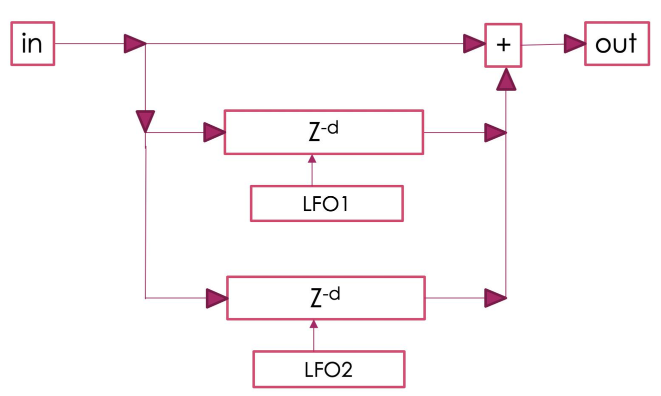
Introduction

Flanger



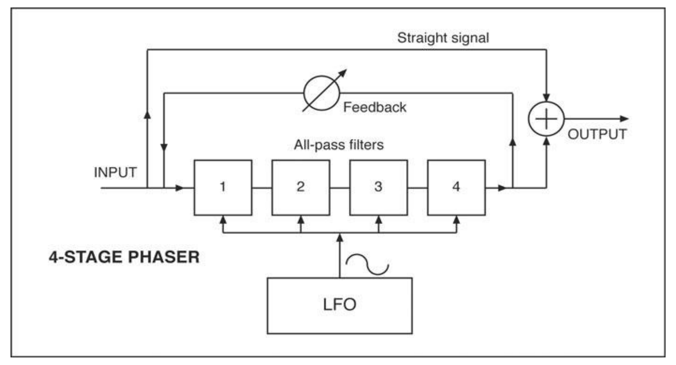
- Variable length fractional delay line modulated by an LFO
- Portion of delayed signal is fed back through delay line
- Delayed signal mixed with dry signal
- Output signal is a comb-filtered version of input signal, with peaks and troughs that are in a linear harmonic series

Chorus



- Signal is split and sent through two variable length fractional delay lines modulated by two independent LFOs
- Delayed signals mixed with dry signal
- Output signal sounds like multiple "voices" of the input signal, resulting in a wider and thicker sound

Phaser



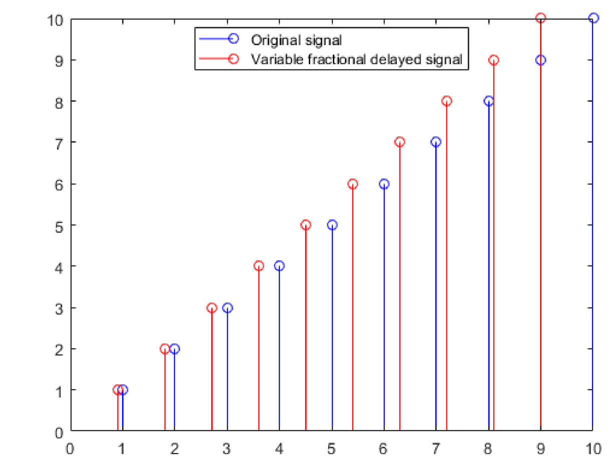
- Signal sent through cascaded allpass filters,
- Phase shift produced by each allpass filter is modulated by an LFO
- Portion of phase-shifted signal is fed back through cascaded allpass filters
- Phase-shifted signal mixed with dry signal
- Output signal is a comb-filtered version of input signal, with peaks and troughs that are not harmonically related

Methods

LFO

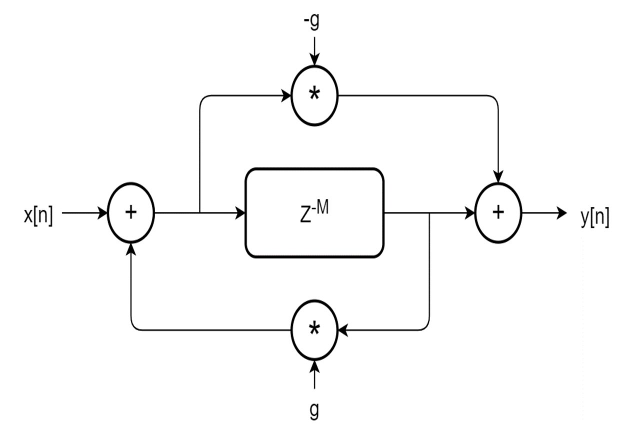
- Used to modulate the length of the variable fractional delay lines (i.e. delay time) and the all-pass filter 'cutoff' frequency
- Implemented as a class in MATLAB so the audioPlugin class for each audio effect can easily reference it
- LFO frequency can be varied from 0 to 10 Hz

Variable Fractional Delay Line



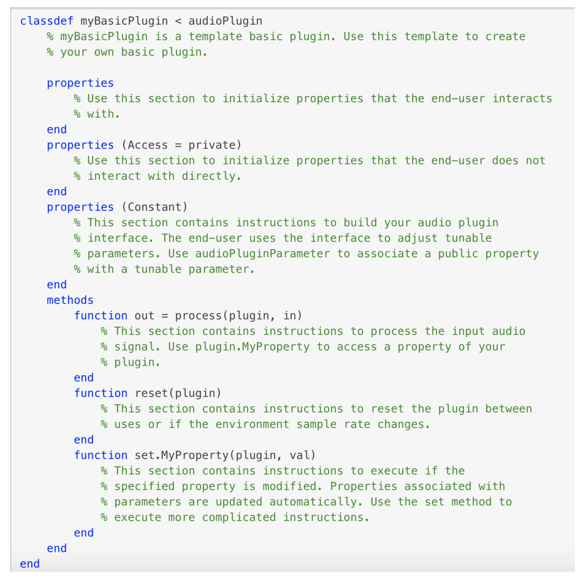
- The basis for all three effects
- Implemented using the MATLAB system object *dsp.VariableFractionalDelay* and *audioexample.DelayFilter*
- Delay value can vary from sample to sample within a frame
- Uses linear interpolation to calculate new samples at non-integer sampling intervals

Allpass Filter



- Implemented by calculating the bi-quad coefficients of the filter
- Passes all frequencies at unity gain (i.e. neither increases or decreases the level of any frequency)
- Alters the phase response of the input signal
- Output signal is a phase-shifted version of the input signal

Generating Real-Time Audio Effects in MATLAB



Results

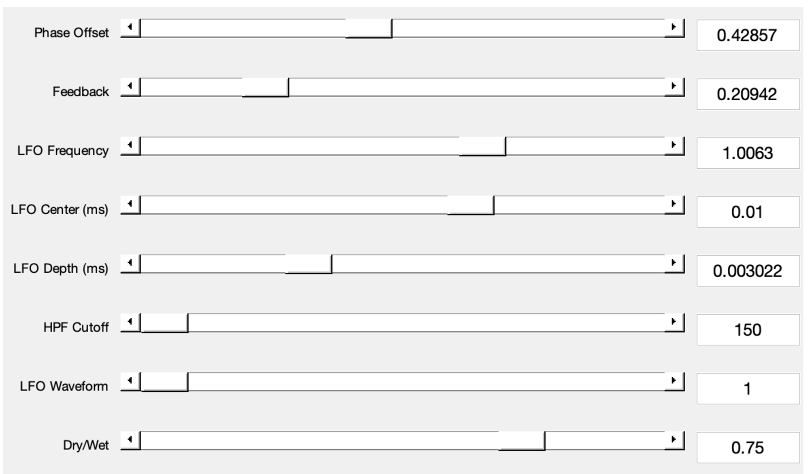


Figure A: Flanger settings

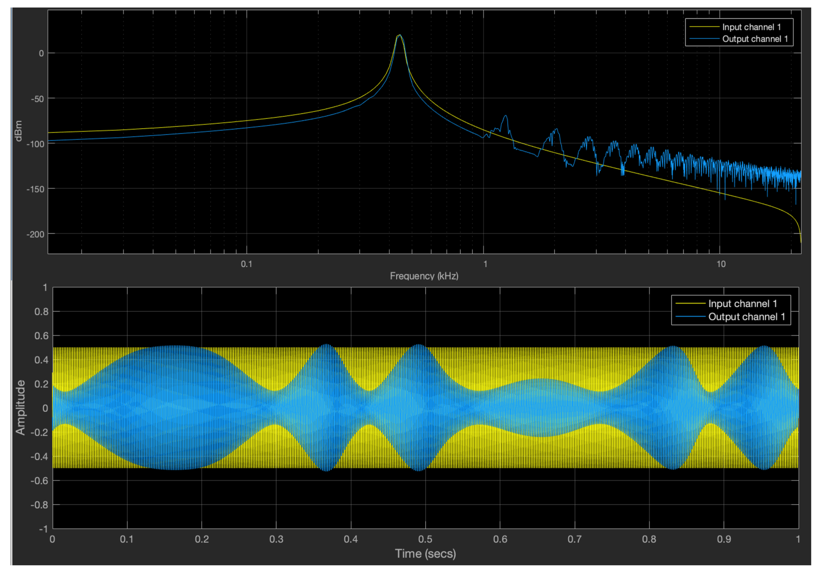


Figure B:
Flanger input and output frequency and waveform display

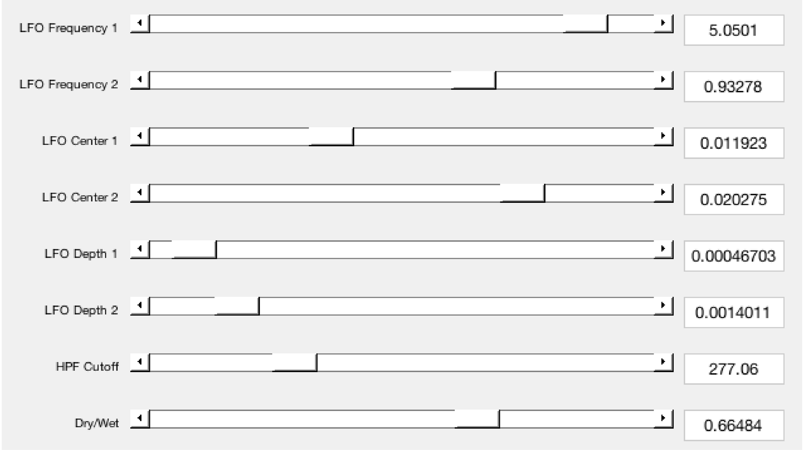


Figure C: Chorus settings

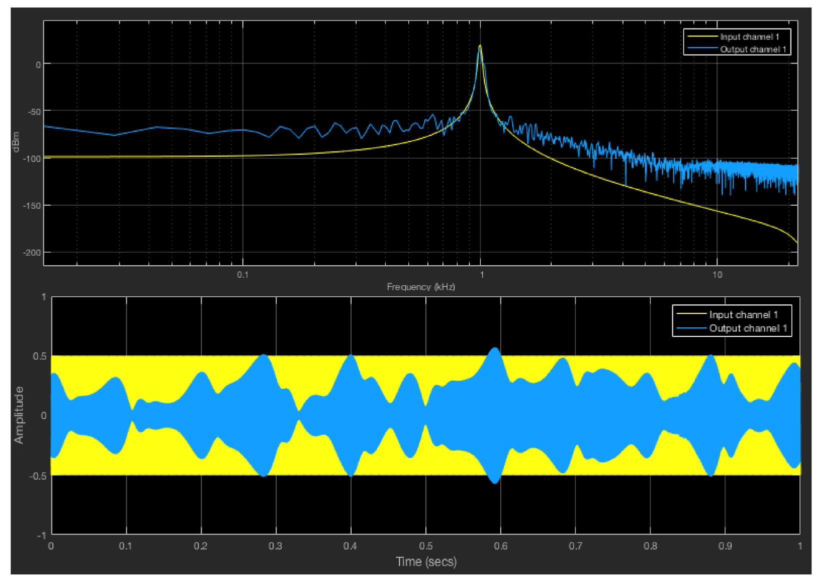


Figure D:
Chorus input and output frequency and waveform display

Results (continued)

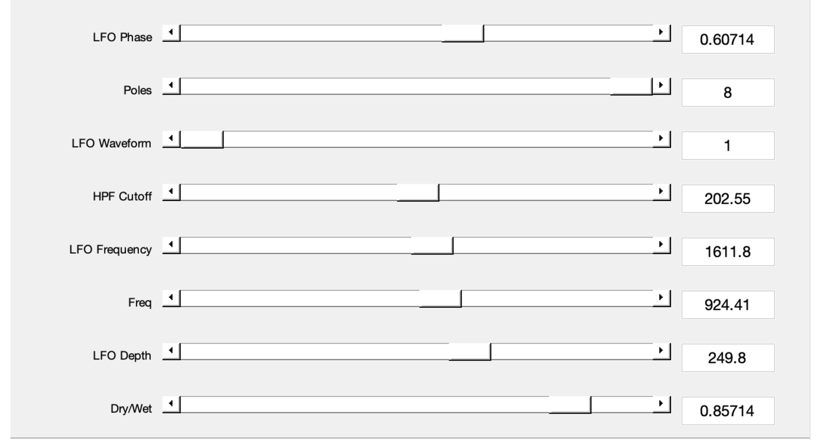


Figure E: Phaser settings

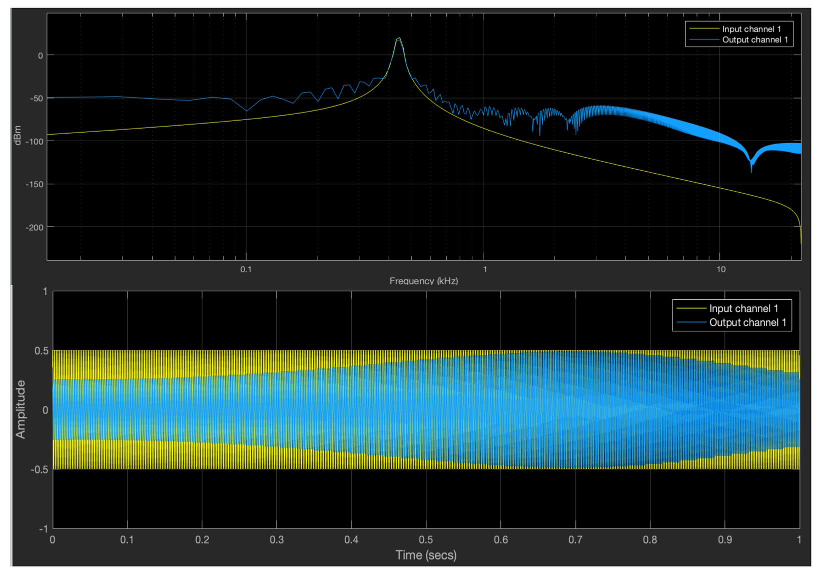


Figure F:
Phaser input and output frequency and waveform display

Discussion

- Each audio effect was tested by inputting a 1kHz sine wave with an amplitude of 0.5
- The frequency spectrum in Figure B shows that the output signal is a comb-filtered version of the input signal, with peaks and troughs that are harmonically related
- The waveform display in Figure D illustrates the randomly modulated delay produced by the chorus effect
- The frequency spectrum in Figure F shows that the output signal is a comb-filtered version of the input signal, though the peaks and troughs are not harmonically related
- Next step in project is to implement functions as Virtual Software Instruments (VST) so they can be used in a DAW

References

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Howell, Steve. "Q. What's the Difference between Phasing and Flanging?" *Soundonsound.com*, SOS Publications Group, Mar. 2006, www.soundonsound.com/sound-advice/q-whats-difference-between-phasing-and-flanging.