FINAL REVIEW - FOUR INPUT AMPLITUDE MODULATOR

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ABSTRACT

This report presents the requirements and assessment of the various processing necessary for an audio effect described here as the four input amplitude modulator. An overview, accompanied by description of the implementation and the audible effects of amplitude and frequency modulation is given. The reader is provided with matlab and audio files that demonstrate the concepts presented.

1. INTRODUCTION

This report describes a matlab implementation of an amplitude modulator similar to the 'Rotor' effect found on the John Bowen Solaris digital synthesizer [1]. The rotor effect can be thought of as a four input amplitude modulator, where the output of the effect is made up of sequential time segments of each input. Frequency (or pitch) modulation (FM) will be applied to increase and decrease the rate of amplitude modulation and the user is also able to select the amount crossfade between amplitude envelopes.

To evaluate this processor, the tonal characteristics are assessed and demonstrated with audio examples. When operating in the audio range, the processor blends the characteristics of the input signal and adds spectral content depending on the crossfade between amplitude envelopes and the rate of modulation. At sub-audio frequencies the processor can be thought of similar to tremolo between four inputs.

2. AMPLITUDE MODULATION

2.1. Amplitude modulation - Concept

Amplitude modulation (AM) is the process whereby a signal has its amplitude varied by the amplitude of a modulating signal (usually a low frequency oscillator or LFO). AM is simple multiplication of the the input audio and modulation signal given by

$$y(n) = [1+\alpha m(n)]*x(n) \quad (1)$$

where the signal is unipolar modulating between 0 and 1. The coefficient α can be thought of as a depth control for the modulator. [2]

In musical applications of amplitude modulation, commonly refer to as tremolo, the subtle modulation of amplitude in the range of 4 – 7 Hz[2], the effect of which is heard as a temporal change of amplitude on the input signal. Considering a carrier wave (f_c) and modulator (f_x) that are both sinusoidal signals in the audio range (f_x >20 Hz), the output will consist of three spectrum components and in this case will be heard as a spectral rather than temporal effect. The frequencies heard are the carrier as well as the sum and difference of the carrier and modulator ($f_c - f_x$, f_c, $f_c + f_x$)[2] or rather, as the the upper side band (USB) and lower side band (LSB) respectively. When a bipolar signal is used, as is the case in ring modulation, the original spectrum is removed and the resulting output contains just the LSB and USB. In the case of this matlab function, the input spectrum is desired in the output audio signal.

(a)

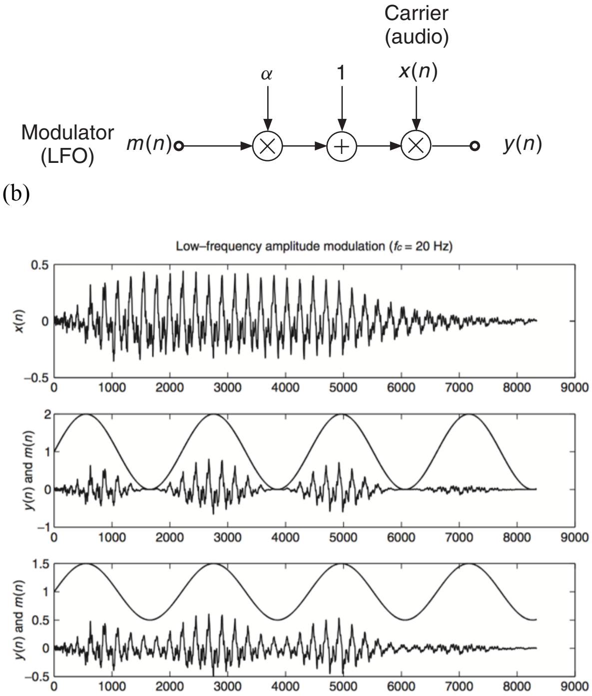

Figure 1. (a) *Typical implantation of Amplitude Modulation (*b) *Example of a tremelo effect from amplitude modulation.[2]*

2.2. Amplitude modulation - implementation

The amplitude modulation frequency, f_x, describes the number of cycles through all four inputs per second, with a useful range of DC to 20 kHz. This processor creates amplitude modulation envelopes similarly to simple noise gates [2]. The fades in and out, hold time and silence are continuously variable to allow the user to modulate the amount of crossfade between envelopes. Figure 2 demonstrates a 1 Hz modulation rate with no crossfade. Each envelope is at an amplitude of 1 for 0.25s per 1 second cycle. The audio inputs 1-4 are sinusoids oscillating at 10 Hz, 20 Hz, 30 Hz and 40 Hz respectively to clearly show the effect of the amplitude envelope on each input.

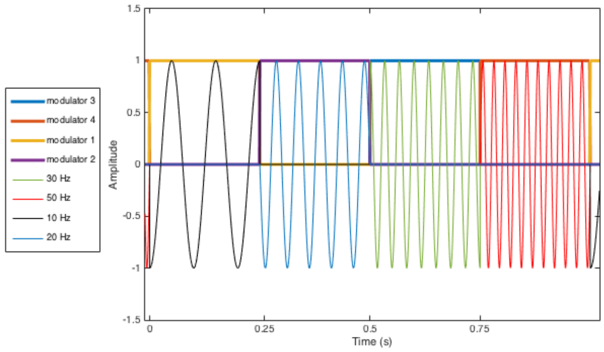

Figure 2. *Example demonstrating modulation of inputs. Thick coloured lines are modulation envelopes 1-4. A square envelope is used to clearly demonstrate the separation of input waves 1- 4. Wave 1 is 10 Hz (thin black line), wave 2 is 20 Hz (thin blue line), wave 3 is 30 Hz (thin green line) and wave 4 is 40 Hz (thin red line).*

The amount of crossfade between envelopes is user controlled and can be a static or modulated variable. A sinusoidal signal can be used to modulate the crossfade which in turn varies the values for the fade and hold time variables and is demonstrated in figure 3.

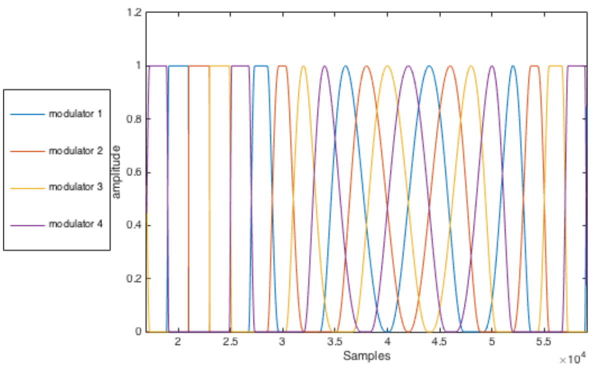

Figure 3. *An example of amplitude envelopes with increasing then decreasing crossfade. Notice the even spacing of amplitude peak centers. The modulation frequency is set at 12 Hz, crossfade modulation is set at 2.4 Hz at maximum depth.*

3. FREQUENCY MODULATION

3.1. Frequency Modulation

It may be of interest to the user to modulate the frequency (f_x) of the modulation envelopes. An example of frequency modulation can be seen in figures 4 and 5. A bipolar sine wave in this case is modulating the frequency of the of a carrier signal. The amplitude of the modulating frequency will determine the maximum change in frequency above or below the carrier frequency. The cycle, from which the fades, hold time and silence are calculated is continuously varied for each sample processed to result in a speeding up and slowing down of the rate of amplitude modulation. This was found to be computationally less expensive than delay line modulation suggested in [4]. Delay line modulation has issues which arise due to the interpolation of the signal as the delay may be a fraction of a sample. [3]. The correction for this is computationally expensive, and increases as the width in Hz increases. While the range of frequency change might be small for a vibrato emulation, the user of the input modulator may have use for 'unnatural' amounts of frequency modulation and a more efficient solution was required.

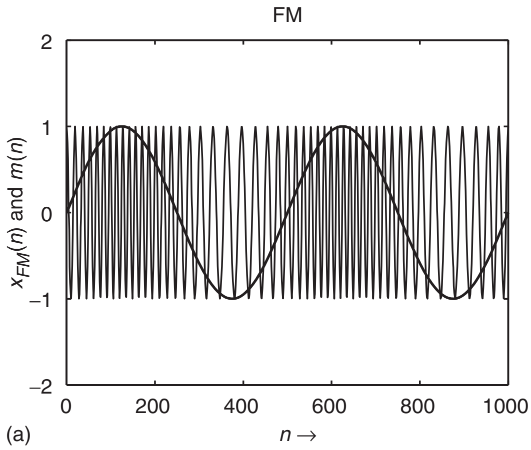

Figure 4. *Carrier frequency modulated by the amplitude of a low frequency sinusoidal signal[2]*

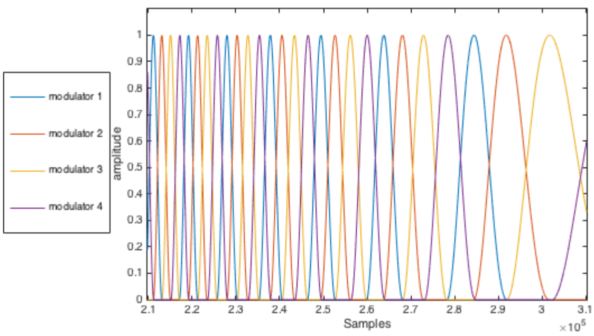

Figure 5. *Four modulation envelopes decreasing in frequency with a constant crossfade.*

4. TONAL CHARACTERISTICS - EVALUATION.

The ability to modify the cross fade amount is essential to the character sound of the output. At a high modulation rate (audio range), the user should be able to hear the change in crossfade as a tonal change. As the crossfade moves from 200% to 0%, there is a noticeable increase in the high frequencies. Figure 6 (sound example – '1000Hz.wav') displays the spectrogram of four octave spaced sine tones (125 Hz – 1000 Hz) with no processing and the four tones are clearly identified. In figure 7 (sound example – '1000Hz_crossmod.wav') the same input tones are processed with crossfade modulation. As the crossfade decreases there is an increase in upper sideband frequencies and may be heard by the user as a low pass filter with an increasing passband frequency as the spectrum becomes more complex.

In contrast, applying frequency modulation results in distinct pitches increasing and decreasing as the rate of modulation varies. This is demonstrated in figure 8 (sound example – '1000Hz_FMmod.wav). Furthermore, it is apparent that new pitches are introduced to the output signal and are related to the input pitches as discussed in section **2.1**.

At modulation frequencies in the audio range it is apparent that this processor adds spectrum components not contained in the input audio and is therefore a useful tone shaping tool for audio engineers and electronic musicians.

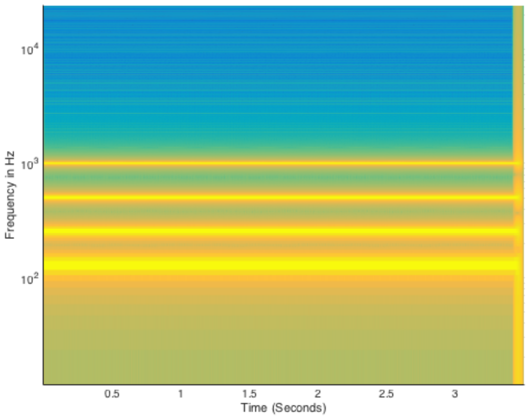

Figure 6. *Spectrogram for octave spaced tones (125 Hz – 1000 Hz) with no processing. (sound example – '1000Hz.wav')*

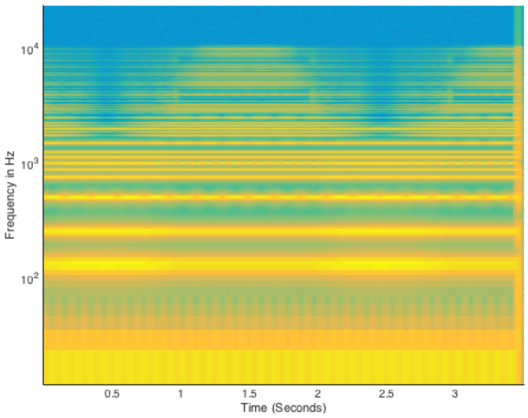

Figure 7. *Spectrogram for octave spaced tones (125 Hz – 1000 Hz) modulating at 1000 Hz with crossfade modulation. Notice the increase in high frequency components as the crossfade decreases. (sound example '1000Hz_crossmod.wav')*

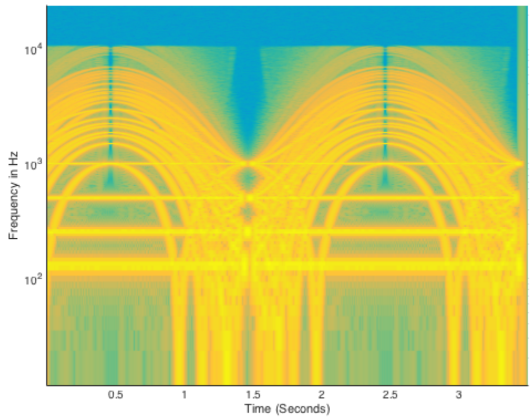

Figure 8. *Spectrogram for octave spaced tones (125 Hz – 1000 Hz) modulating at 1000 Hz with frequency modulation. Notice the high frequency components rising as modulation frequency increases. (sound example '1000Hz_FMmod.wav')*

When the modulation rate is below the audible range of human hearing, the output will be heard as a slow fading or sharp cut between inputs with crossfade and no crossfade respectively. Figure 9 displays the four sine tone inputs modulated at 0.1Hz and a crossfade of 100%[1]. In this case each input fades into the first 50% of time for the following input.

This is a useful tool particularly when different parameters are used for the Left and Right channels. Slow modulation rate and large crossfades produce a slowly evolving mixing between inputs. (sound file – 'monks.wav')[5]. Additionally, the user may use the processor to cut between various drum loops as inputs with less crossfade for interesting effect.

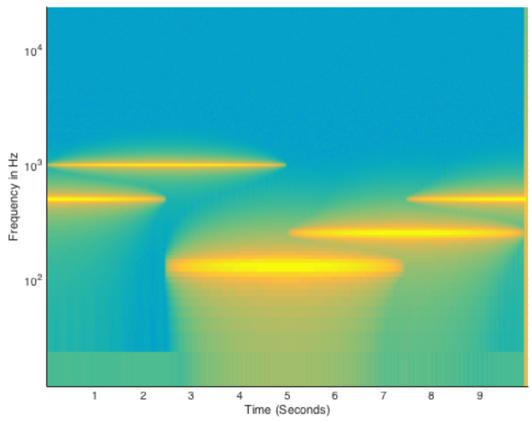

Figure 9. *Spectrogram for octave spaced tones (125 Hz – 1000 Hz) modulating at 0.1 Hz and crossfade depth 100%. (sound example 'subaudio.wav')*

5. USER INTERFACE

As simple user interface is provided in order to layout the many input parameters in a logical, easy to use fashion (figure 10). The interface takes user defined audio files or allows for sine tones to be produced as inputs. The sampling frequency multiplier is useful when frequency modulation is used at high audio range

[1] Crossfade depth range (0 – 200%). At 200% the envelope occurs across the length of 1 cycle.

amplitude modulation frequencies. At high frequencies the frequency modulation is not smooth due to low resolution in the FM oscillator and this parameter accounts for this.

The interface allows the user to replay the processed audio and then save the audio file to disc when satisfied with the result.

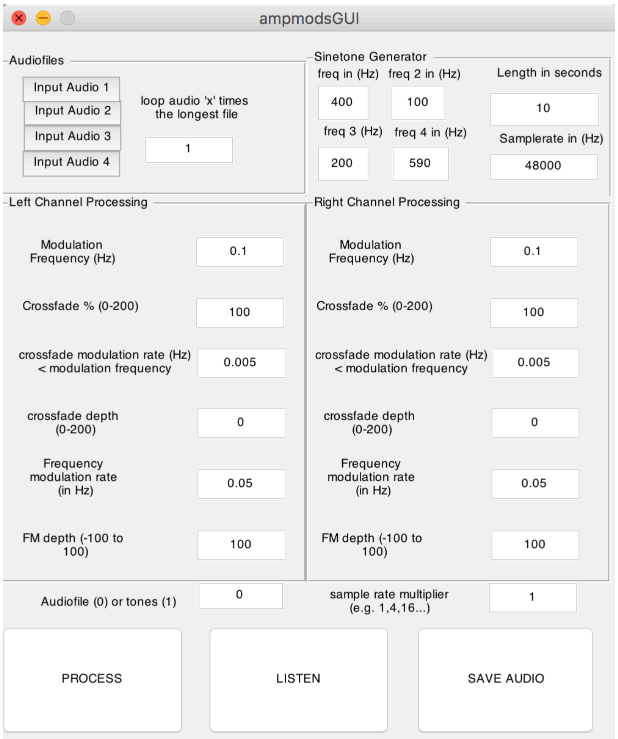

Figure 10. *User interface for the input modulator.*

6. CONCLUSION

Frequency modulation and various methods for amplitude modulation have been discussed in relation to an implementation of a four input modulator. Graphical representations have been presented to show the processor is functioning as intended and effect on the audio for a human listener has been discussed. The amplitude modulator has a wide range of tonal possibilities and these have been assessed and demonstrated with audio examples provided.

7. REFERENCES

[1] https://www.youtube.com/watch?v=_SfhVmRiQB4

[2] U. Zolzer, (2011). *Modulators and Demodulators,* in DAFX: Digital Audio Effects, John Willey & Sons, pp. 93 – 94.

[3] Dattorro, J. (1997). Effect design, Part 2: Delay line modulation and chorus. Journal of the Audio engineering Society, 45(10), 764-788.

[4] S. Dirsch, U, Zolzer, (1999). Modulation and delay line based digital audio effects, Proceedings of the 2nd COST G-6 Workshop on Digital Audio Effects (DAFx99), NTNU, Trondheim.

[5] Original sound files from http://ccmixter.org/files/ztutz/2362