Digital Audio Interface Design

Module Code: MU70004E

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A Harmonic Saturation Unit

Introduction

The main stages of this unit will involve using Fast Fourier Transforms, audio analysis and synthesis. Essentially the patch will receive audio and analyse it, the analytics can then be used to locate the lowest harmonic also known as the fundamental. This could then be tracked and output to an oscillator bank in order to build a harmonic profile from the source signals fundamental. This profile or energy weighting would then be applied back to the source in order to give a saturation effect. An alternative to using the oscillators would be to track the first order harmonics present in the lowest octave and using them to proportionally alter higher bands.

Background

This project stems from an interest in subjective views of audio distortion and our perception of warmth. In various articles audio professionals comment on the 'warmth' of tube equipment, Russell O. Hamms[1] paper '*Tubes Versus Transistors; Is There an Audible Difference*' addresses how a harmonic profile correlates to our perception of an instruments timbre. In this paper its shown that tubes exhibit a stronger second harmonic the further they are driven into their distortion range; in the musical realm this second harmonic is one octave higher than the lowest note. This is commonly used to 'solidify' the bass note and perceptually could be seen as a supporting note that adds an indistinct second tone.[2] As Hamm states:

'Musically the second is an octave above the fundamental and is almost inaudible; yet it adds body to the sound, making it fuller.'[3]

In the audio industry there has been a recent rise in the availability and popularity of saturation units; its the authors opinion that this is because harmonic saturation is an unobtrusive way to perceptually change the 'colour' or 'impression' of a source signal.[4]

With this in mind the aim of the patch is to create a unit with variable parameters which can be manipulated in order to musically alter a source signals 'colour', this will be achieved through the application of weightings to various harmonics present in a complex signal.

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1 Hamm, R. (1973) - *Tubes Versus Transistors; Is There an Audible Difference* – Audio Engineering Society.
2 Think of the left hand playing the bass on a piano, if the line is uncomplicated it is common to play as octaves. Similarly on a guitar the root note and the octave higher is played within many chords.
3 Hamm, R. (1973) p6 - *Tubes Versus Transistors; Is There an Audible Difference* – Audio Engineering Society.
4 For instance the recent release of AVIDs 'Heat', the option to add harmonics in 'Smack!' or their 'Reel Tape Suite'.

The Patch
In order to build a harmonic profile its necessary to have a fundamental frequency to begin with. To find this frequency we apply FFT and track the loudest sinusoid in a signal, this tracking can be done with an external object called centroid~.[5]

The output of centroid~ is the fundamental and can be fed to several different multipliers, the amount of times multiplied would then be proportional to the harmonic. Each of these in turn can be fed into an oscillator and summed the result is a harmonic series. However in order to emulate various types of distortion the amplitude of each oscillator must work independently, this would allow a user to build their own harmonic series that stays tracked to the 'present' fundamental.

With the oscillator bank built the next step is to apply the tracked harmonic series to our audio, however we don't want to combine the oscillator banks signal with the source, we want to apply its energy weighting. To do this would require running two FFTs in parallel; one for the source and one for the oscillator bank. The FFTs could then be convoluted onto each other, with an Inverse FFT translating this back to the audio domain.

The Fast Fourier Transform has an inherent trade off between frequency and timing resolution; as the FFT size is increased the temporal resolution is reduced. To make this system more accurate two FFTs can be run in parallel with an offset between them, each of these FFTs are 'enveloped' to create a cross over between them, this can be done semi-automatically with the pfft~ object; simplifying things while also being more efficient.

When the two signals reach the pfft~ sub patcher object, or more precisely the fftin~ object, they can be convoluted. To perform this convolution remapping each of the two FFT signals from cartesian numbers to polar coordinates is necessary, this can be achieved with the cartopol~ object; giving out the relative amplitudes and phases of each frequency. We can then multiply the amplitudes and add the phases of each FFT signal in order to superimpose them onto each other. We then remap back to Cartesian numbers before using an Inverse FFT which completes the convolution process. The input signal should now still be recognisable, ideally with a subtle 'colour' added.

Changes
Although this approach seemed to work in theory I have had difficulty achieving the desired results. Firstly, through the scope~ object, I have observed that the oscillators don't remain phase locked as the centroid~ object inputs its data. I attempted to remedy this problem by using the phasor~ to sync them but to no avail.

A more critical realisation I've had about the design is that it that it only accounts for a single frequency and the relevant harmonics. This is not in keeping with any amplifier design where the harmonics created by the device are proportional to incoming frequency energy across the spectrum; each frequency in the first octave has a proportional amount of energy created in the second, third, etc.

5 The loudest frequency will be the fundamental as it contains the most energy.

To work like this would perhaps require a different approach; given that we can analyse the frequencies present in a signal, their energy weightings with an FFT, and also knowing the FFT 'bin' width it should be possible to calculate the harmonics of each band. It should be noted that although the bins are octaves/harmonics apart the frequencies are unlikely to fall exactly on the FFT frequency. If this is correct it should also be possible to distribute extra energy on a proportional basis related to the fundamental frequency band, by a user defined amount.

To work out exactly how to modify the FFT data in this way will require a deeper understanding of the FFT process and the MSP objects I can use to manipulate its signal.

One possibility I've come across could be using the table/itable object; this can generate transfer functions which could be 'used to weight the output based on the input'.[6] Here it would be a matter of mathematically altering the output of the second octave by a proportional amount to the amplitude in the first octave. To add any further harmonics you would vary the other octaves by a proportional amount to the first.

User interface

Ideally the user will have controls to increase/decrease chosen harmonics and a master saturation knob for ease of use, this will essentially be a mix knob. Presets could be created and made easily accessible for different distortion characteristics typical of various units.

Summary

This unit will work by applying FFT and modifying the signal to amplify the harmonics present. This will be a challenge given my limited knowledge of maths, physics and MaxMSP; however it is a project worth pursuing because its an opportunity to learn about the DSP happening below the typical user interfaces. Given that recording and mixing are nearly completely digital now, knowledge of the underlying DSP can help when making critical decisions about processing audio at the user level. It has also demonstrated that nearly all audio processing requires tradeoffs between the integrity of the source signal, the artefacts introduced into the signal and the perceptive benefits of the signal processing.

Word Count: 1194

6 Cycling '74 (2010) - Max Basic Tutorial 17: Data Structures and Probability – MaxMSP v5.1.5 software documentation.

Proposed work schedule

Week 7
Further research into the broad subject of FFT
Building project

Week 8
Research into frequency analysis
Building project

Week 9
Research into frequency manipulation
Building project

Week 10
Research into working with data tables to store transfer functions
Building project

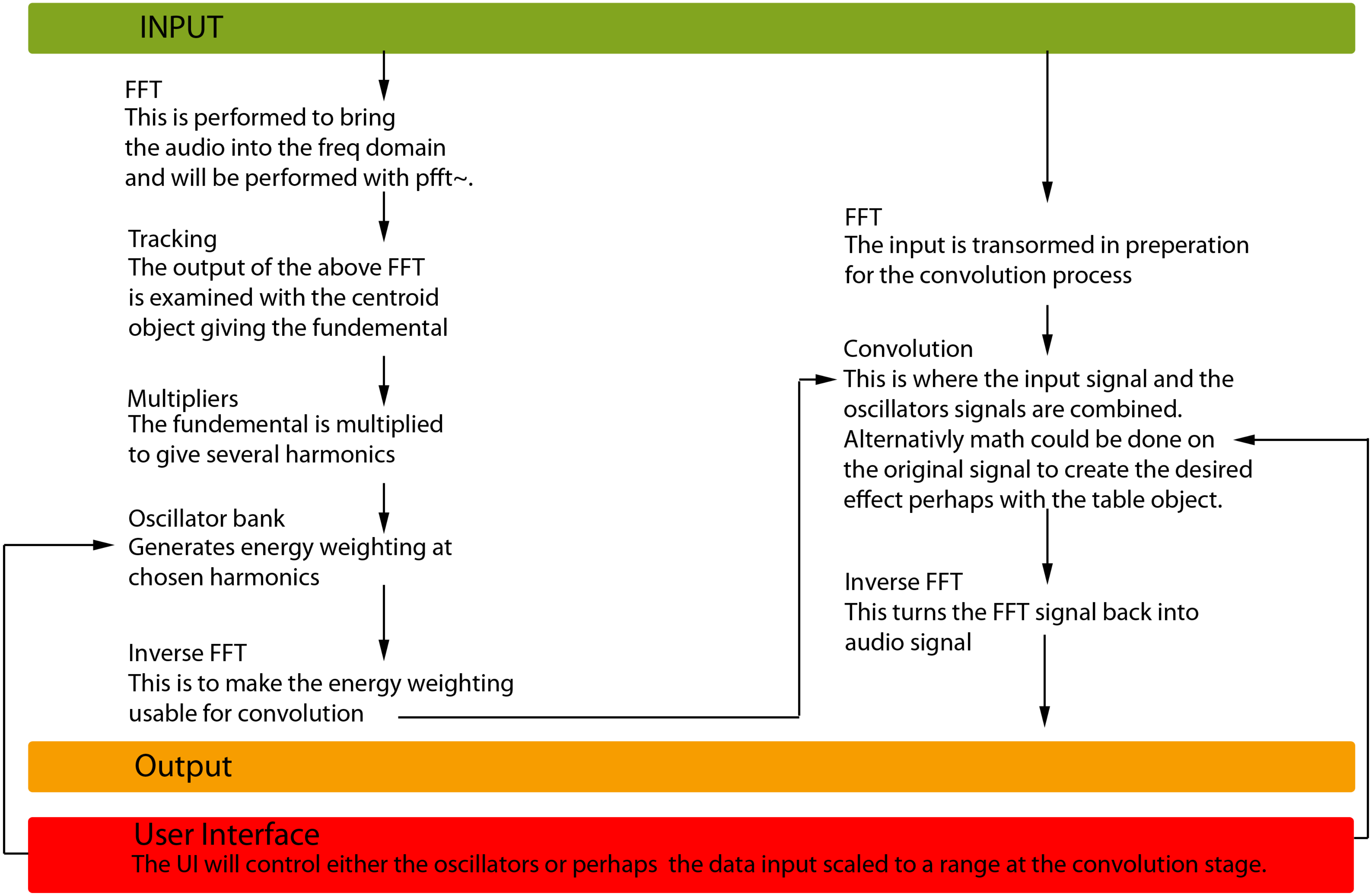
Week 11
Further research into the various areas mentioned
Building project

Week 12
Further research into the various areas mentioned as well as designing the interface, dependent on the strategies employed it may be necessary to work on linking to control surface in order to comply with the 'three subjects' criteria
Written work
Building project

Week 13
Further research into the various areas mentioned
Written work
Building project

Week 14
Project hand in

Flow Chart


INPUT
FFT
This is performed to bring
the audio into the freq domain
and will be performed with pfft~.
Tracking
The output of the above FFT
is examined with the centroid
object giving the fundemental
Multipliers
The fundemental is multiplied
to give several harmonics
Oscillator bank
Generates energy weighting at
chosen harmonics
Inverse FFT
This is to make the energy weighting
usable for convolution
FFT
The input is transormed in preperation
for the convolution process
Convolution
This is where the input signal and the
oscillators signals are combined.
Alternativly math could be done on
the original signal to create the desired
effect perhaps with the table object.
Inverse FFT
This turns the FFT signal back into
audio signal
Output
User Interface
The UI will control either the oscillators or perhaps the data input scaled to a range at the convolution stage.

Comment List

#	Page #	Symbol	Content
1	4		*It might be an interesting approach to test the result from centroid~ with pitch information generated by fiddle~, analyzer~ or sigmund~*
2	4		*At places like this your argument would be enhanced by stating conversion algorithms, to show your knowledge of them.*
3	4		*Any more detailed approach researched on what you'll be using for the oscillator banks? various different approaches are possible here, from multiple cycles, to very sophisticated systems such as the resonator~*
4	5		*This is a very powerful approach, showing your very informed research into the subject! Particularly as I would not expect such kind of critical evaluation in a proposal! I would suggest that this approach will be the more useful one, although the previously described mode might have very powerful applications as well for monophonic instruments/tracks. Therefore both approaches might be worth considering as features?!*

General Comments

This is a very convincing proposal showing very informed statements based on a well researched topic. Credits have been given for a very informed critical evaluation of your approaches in this early stage of the project. I would also suggest to consider alternative strategies to enhance the audio other than convolution. You might find the audio results from fft data controlled filters (biquad~, cascade~, fffb~) aesthetically more pleasing, by avoiding a fairly typical fft sound.

Some more details on the proposed user interface would have made your proposal even stringer, particularly as you seem to have a very clear vision what the user might want/need to control for a successful application of this process.

Although you have given references in your footnotes, please make sure you include a bibliography in your essays!