Electronic Music Synthesis and Audio Effects Processing

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Abstract— Music is a gratifying part of the life of a plethora of people in the world. The application of various signal processing techniques in the field of music has paved way to 'Music Technology'. Music technology has changed the way music is created, composed, stored and analysed. A user friendly easy to use GUI is proposed to be designed on MATLAB to facilitate creation of music using specially synthesized virtual digital musical instruments. The creation of music is proposed to be achieved using Physical Modelling Techniques. In addition to composing various patterns using the instrument models, inclusion of various sound effects using Audio Signal Processing techniques is also proposed.

Keywords- Audio effects, Music synthesis, Matlab, Digital audio technology, Additive audio synthesis, FM audio synthesis.

I. INTRODUCTION

Virtual Musical Instrument is the new trend of the digital era. Virtual Instruments are useful in the synthesizing, recording, manipulation, mass-production, and distribution of sound. Various audio players have been designed where the user can play these audio files but with certain limitations such as fixed tempo, fixed scale and not so friendly user interfaces. There is a need of modern design and implementation of virtual instruments that are more user friendly and offer more features that can manipulate music to suite the taste of individual human beings.

Our work has three aspects. Firstly, the work is focused on providing instrument sounds which can be used to synthesize music. These virtual instruments are proposed to be implemented in the user friendly MATLAB GUI. This allows users to interact with electronic devices through graphical icons and visual indicators, as opposed to text-based interfaces. Using this GUI, a user can access various audio files stored on his device. The next aspect of the work was enabling the user to play his synthesized audio wav files on his device by applying certain effects that will enhance his listening experience. The effects such as Flange, Base, Treble, Modulation, Wah Wah effect are implemented using Matlab and these are hence linked to the GUI. While playing the musical notes Musicians can make mistakes but the computer program will be very accurate. Next aspect of work focused on inputting notes to the software will generate the music accordingly.

A. *Musical notes, Pitch and Octave*

Music and noise are both mixtures of sound waves of different frequencies. [1] Music is characterized by the presence of 'pure tone' i.e. the dominant frequency, called as the 'fundamental' and its ratios, called overtones. Noise is characterized by the presence of a continuous range of frequencies, with no discernable dominant frequency. Thus it can be said that music is a sound with 'harmony' between the entailing frequency components and that the presence of harmonious frequency components is what creates the pleasing effect. Musical note is the fundamental element of music. A note is a pitched sound i.e. sound corresponding to a particular frequency within the audible range (20 Hz to 20 kHz). The pitch is that aspect of sound which is perceived as the 'sharpness' of the sound. Pitch is related to the frequency of sound, but it is a subjective term. The note with a higher frequency is said to have a higher pitch and the one with lower frequency is said to have a lower pitch. The frequencies of the successive notes are related by a fixed ratio and it is found that the frequency doubles over a set of 12 consecutive notes.

Thus $(2f) = (x^{12}) * (f) \qquad ,X=1.0594$

This set of 12 notes over which the frequency gets doubles is known as octave [2]. In western music the notes in an octave are named as follows: C, C#, D, D#, E, F, F#, G, G#, A, A#, B, C (the last C corresponding to the next octave).

TABLE I. LIST OF FREQUENCIES OF THE NOTES IN HERTZ OVER 9 OCTAVES

	C	**C#**	**D**	**D#**	**E**	**F**	**F#**	**G**	**G#**	**A**	**A#**	**B**
0	16.35	17.32	18.35	19.45	20.60	21.83	23.12	24.50	25.96	27.50	29.14	30.87
1	32.70	34.65	36.71	38.89	41.20	43.65	46.25	49.00	51.91	55.00	58.27	61.74
2	65.41	69.30	73.42	77.78	82.41	87.31	92.50	98.00	103.8	110.0	116.5	123.5
3	130.8	138.6	146.8	155.6	164.8	174.6	185.0	196.0	207.7	220.0	233.1	246.9
4	261.6	277.2	293.7	311.1	329.6	349.2	370.0	392.0	415.3	440.0	466.2	493.9
5	523.3	554.4	587.3	622.3	659.3	698.5	740.0	784.0	830.6	880.0	932.3	987.8
6	1047	1109	1175	1245	1319	1397	1480	1568	1661	1760	1865	1976
7	2093	2217	2349	2489	2637	2794	2960	3136	3322	3520	3729	3951
8	4186	4435	4699	4978	5274	5588	5920	6272	6645	7040	7459	7902

From the above table it can be seen that the frequencies along the column are integral multiples of the previous entries, these frequencies are referred to as "Harmonics". i.e. C2 and C3 are the second and third harmonics of C1 respectively. The successive frequencies along the rows have a ratio of 1.0594; they are referred to as overtones, i.e. all the frequencies post C1 in the second row are the overtones of C1.

B. Timbre and Physical modelling:

In music, timbre (pronounced Tam-ber) also known as tone color or tone quality from is the quality of a musical note, sound, or tone that distinguishes different types of sound production, such as voices and musical instruments, string instruments, wind instruments, and percussion instruments. [3] The instrument sounds can be differentiated even when they are playing the same note with same pitch and with same loudness, the parameter that has differed is the timbre. Thus timbre can be defined as "the characteristic quality of a sound, independent of pitch and loudness, from which its source or manner of production can be inferred". Thus timbre is a subjective term, although various factors affect the timbre it is observed that there are two important affecting timbres. [5]

Timbre can be said to be comprised of Harmonic content or the Frequency Spectrum and Amplitude envelope. In reality whenever a note is produced by an instrument or a human voice, the harmonics and overtones corresponding to that note are also produced along with the fundamental frequency. The relative energy distribution of these partials is said to greatly affect the timbre. The manner in which amplitude of the sound rises and decays i.e. the attack, sustain and decay patterns of sound or the amplitude envelope is also found to affect the timbre. A technique of artificially synthesizing instrument sounds based on the frequency spectrum and amplitude envelope of real instrument sounds is known as physical modeling. This is the technique employed in our work, the instrument sounds have been generated based on the above mentioned timbre parameters.

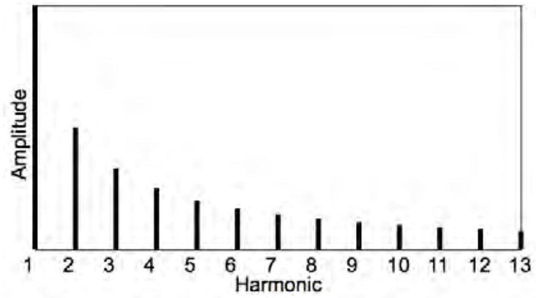

Figure 1. Harmonic content of a sound.[4]

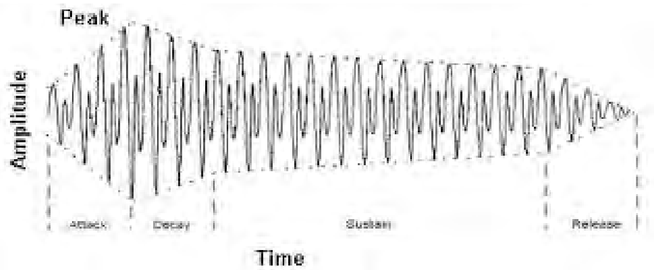

Figure 2. Attack, Decay, Sustain and Release (ADSR) envelope. [7]

II. SYNTHESIS TECHNIQUESADDITIVE SYNTHESIS

A. Additive Synthesis

Additive synthesis is a sound synthesis technique that creates timbre by adding sine waves (or any periodic wave as required) together. This concept of constructing a complex sound out of sinusoidal terms is the basis for additive synthesis, sometimes called Fourier synthesis since it is the reverse application of Fourier theorem i.e. any complex waveform can be generated by summation of sinusoids.

$$f(t) = a_0 + \sum_{n=1}^{\infty} a_n \sin(f_n t)$$ [2]

In addition to this, the concepts of additive synthesis have also existed since the introduction of the organ, where different pipes of varying pitch are combined to create a sound or timbre.

Key factors considered are the total number of sinusoids and the number of harmonics that directly influences the quality of sound. Higher the number of harmonics higher is the quality of sound. A sound which consists of just three sinusoids will most likely sound poorer than one which employs 100. The base frequency or fundamental frequency is the frequency of the first partial. If the partials are exhibiting a harmonic ratio the fundamental frequency will have the highest amount of energy. The frequency ratios of the sine generators: the harmonic content of the sound i.e. which harmonics and overtones of the fundamental are present. [8]

The amplitude ratios of the sinusoids are also very important in determining the resulting timbre of a sound. This decides the relative energy levels of the partials present. If the higher partials are relatively strong, the sound will be perceived as being more 'brilliant' or ‘sharp’, if the higher partials are soft, then the sound will be perceived as being ‘deep’. In our work all the above mentioned parameters are extracted from the frequency spectrum of various instrument sounds.

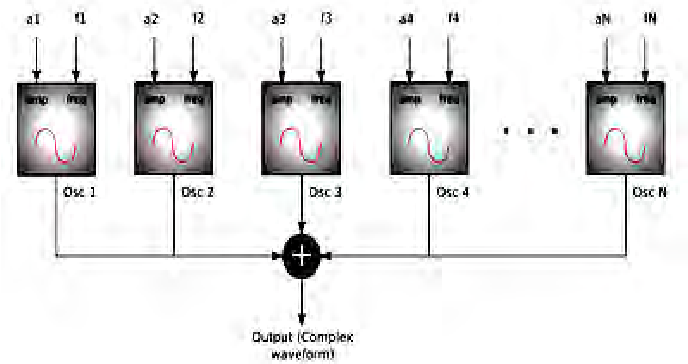

Figure 3. Additive synthesis [2]

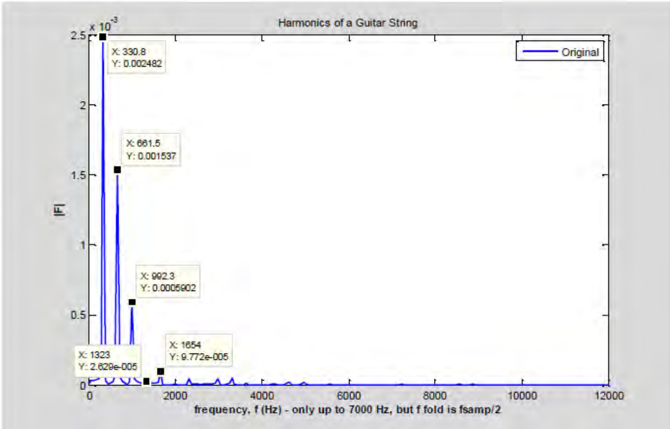

Figure 4. Frequency spectrum on which the synthesis of guitar is based

B. FM Synthesis

Frequency Modulation synthesis has been widely used in music to create special effects such as Vibrato and also in the synthesis of percussion sounds.

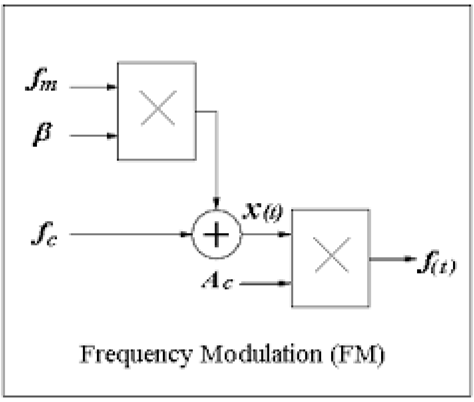

Figure 5: FM Synthesis

F(t)=Ac*cos(2*pi*fc*t+β*cos(2*pi*fm*t)) [2]S

Where F(t) is the frequency modulated signal

Ac and fc are amplitude and frequency of the carrier respectively

β is the modulation index

fm is the frequency of the modulating signal

By suitably adjusting fm and β the vibrato i.e. the wavering of the pitch or the frequency oscillation has been achieved.

III. INSTRUMENTATION

A. Piano

Piano is a struck string instrument which is played with a keyboard. Each key is attached to a hammer which strikes a string when the key is pressed. Sound is produced from the vibrating string and the sound is transmitted to the soundboard.

x=10*sin(2*pi*f*t)+(5*sin(1*pi*2*f*t))+(2*sin(2*pi*3*f*t))+sin(2*pi*4*f*t)+ (0.5*sin(2*pi*5*f*t)); [6]

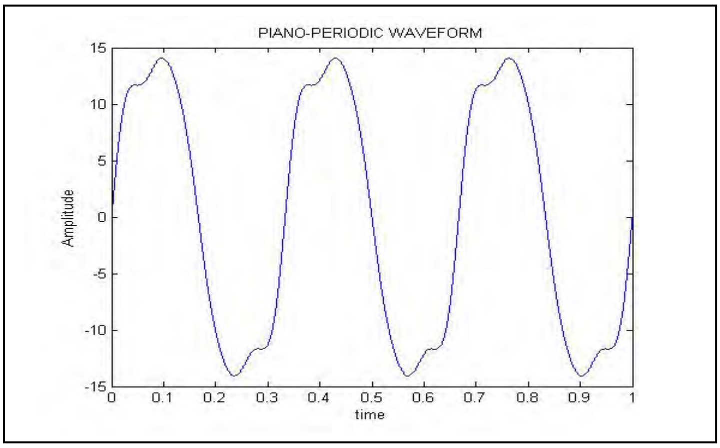

Figure 6: Plot of the Fourier waveform of piano with f=3 [6]

The frequency spectrum of the piano is characterized by the presence of a few harmonics (4 or 5) with comparable energies. Frequency spectrum of Piano with f=523 Hz is shown in below figure 7.

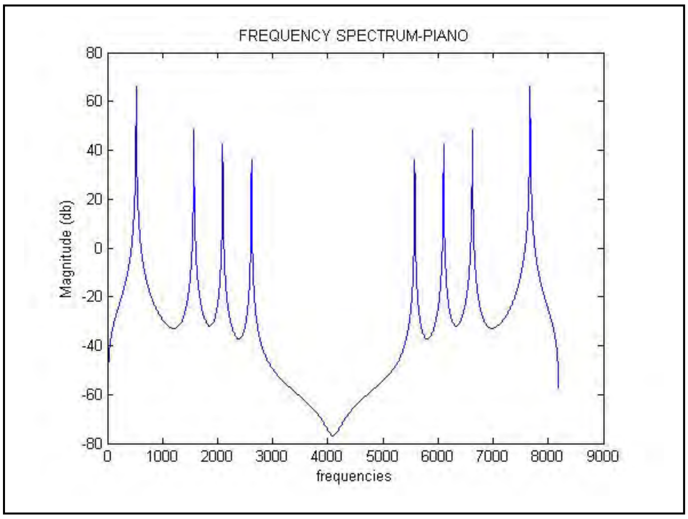

Figure 7: frequency spectrum of Piano. [6]

The amplitude envelope of the piano is an "Attack and Decay" envelope, characterized by fast attack and slow decay as shown in figure 8.

e= (1-exp (-5*t)).*exp (-10*t) [7]

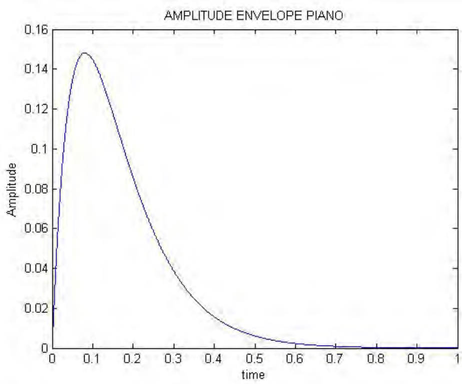

Figure 8: Amplitude Envelope [7]

Thus the overall piano waveform is,

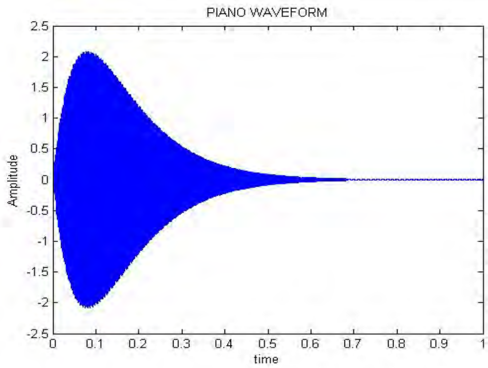

Figure 9: Piano waveform.

B. Flute

Flute is a widely used reed less wind instrument, where sound is produced when air is blown through a blowing hole into to the resonant cavity. The air stream excites the air column in the resonant cavity and produces the sound. [9]

x=1000*cos(2*pi*f*t)+31.6*cos(2*pi*2*f*t)+56.234*cos(2*pi*3*f*t)+17.782*cos(2*pi*4*f*t)+25.1188*cos(2*pi*5*f*t)+15.8489*cos(2*pi*6*f*t)

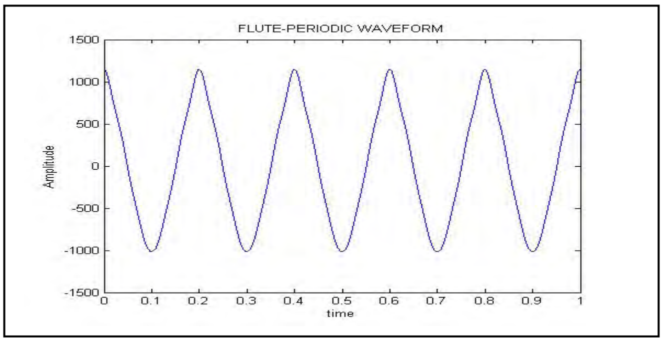

Figure 10: Flute Periodic wave form with f=5.

The frequency spectrum of the flute (Claribel) with f=523 is as shown in the figure 11. It can be observed that most of the acoustic energy is concentrated on the fundamental and that the even harmonics have lesser energy compared to odd harmonics. This gives a hollow texture to the sound. Thus the above flute (Claribel) is more suitable for solo performances [9]. The amplitude envelope of the flute is a monotonically increasing function

e=1-exp(-0.01*t).

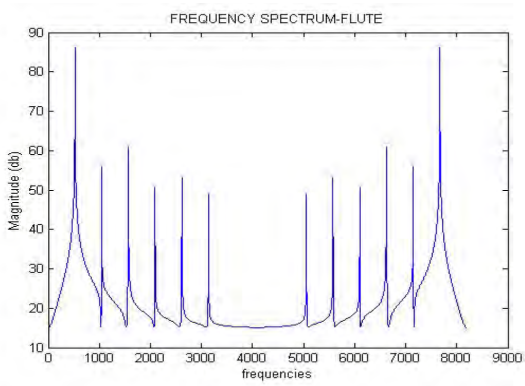

Figure 11: Frequency spectrum of Flute.

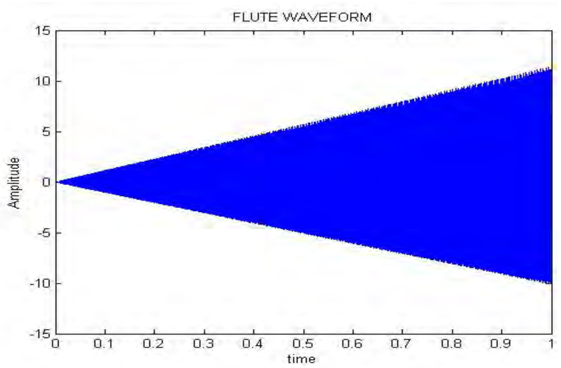

Figure 12: Flute Waveform

C. Clarinet

The clarinet is a family of woodwind instruments that have a single-reed mouthpiece, a straight cylindrical tube with an approximately cylindrical bore, and a flaring bell. The cylindrical bore is what gives clarinet its distinctive timbre. The periodic wave is a sum of sinusoids, based on the frequency spectrum of the clarinet sound.

x=0;

A= 10000*[$10^{(-4.5)}$ e^{-9} $10^{(-9.5)}$ e^{-10} e^{-10} e^{-10} e^{-10} e^{-11} e^{-11} e^{-11} e^{-11} e^{-11} e^{-11} e^{-11} e^{-8} e^{-6} e^{-7} e^{-5} $10^{(-9.5)}$ $10^{(-5.5)}$ e^{-7} e^{-11} e^{-11} $10^{(-9.2)}$ $10^{(-9.8)}$ $10^{(-9.4)}$ $10^{(-10.5)}$];

m=[1 1.18 1.584 1.333 1.779 2.669 3.361 2.244 2.827 3.172 4 4.237 4.75 5.989 2.117 3 3.173 2 1.68 4 5.03 6.346 6.723 7.12 8 9.51 10.07];

```
for i=1:length(A)
  x=x+(A(i)*sin(2*pi*m(i)*f*t));.
end
```

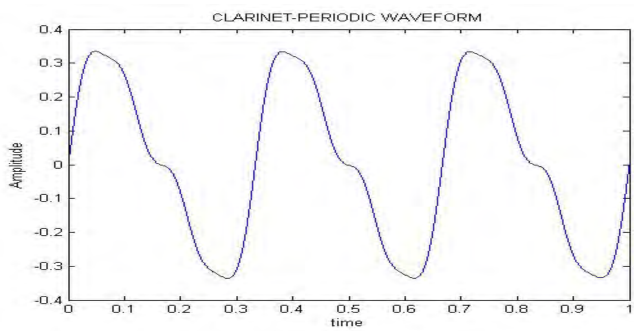

Figure 13: Periodic waveform of clarinet with f=3

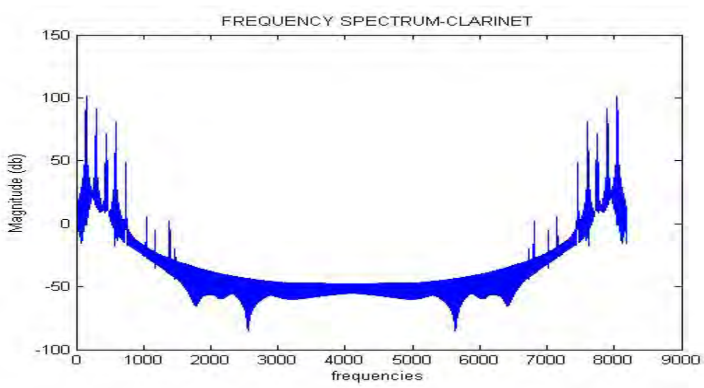

Figure 14: Frequency spectrum of the clarinet.

The frequency spectrum of the clarinet with f=146 is as shown in the figure 14. It can be seen that harmonics are densely present throughout the spectrum and that energy of the harmonics is concentrated more on the edges of the spectrum than on the centre.[10]

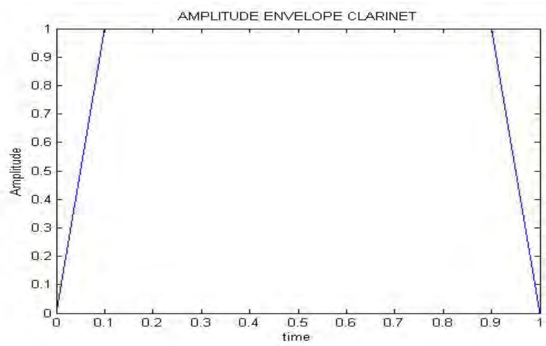

Figure 15: Amplitude envelope of clarinet.

It can be seen that the envelope of clarinet is an ATTACK, DECAY, RELEASE (ASR) envelope.

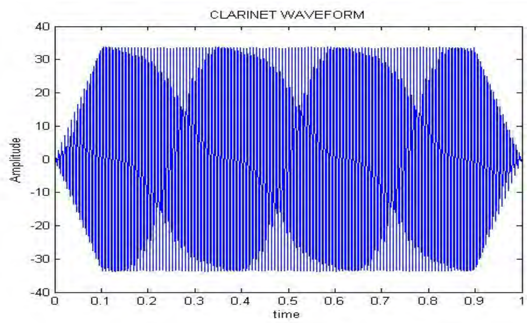

Figure 16: clarinet waveform.

D. Guitar

Guitar is a string instrument which is generally played by strumming or plucking the strings with the right hand and fretting the strings with the left hand. Sound is produced by the vibrating strings and is amplified acoustically or electrically. [11]

BASIC PERIODIC WAVE:

u=248.2*sawtooth (2*pi*f*t,0.5);

v=153.7*sawtooth (2*pi*2*f*t,0.5);

w=59.02*sawtooth (2*pi*3*f*t,0.5);

x=2.629*sawtooth (2*pi*4*f*t,0.5);

y=9.77*sawtooth (2*pi*5*f*t,0.5);

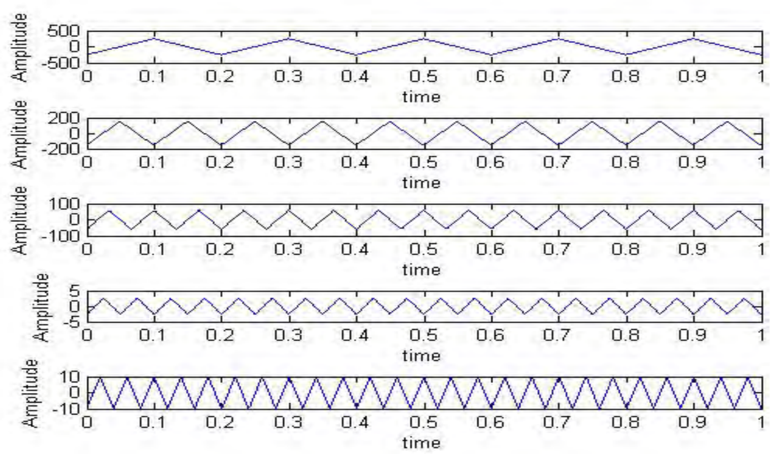

Figure 17: Periodic wave used in Guitar

The figure 18 shows various frequency components of the guitar. It can be seen that the basic periodic wave to be used is the triangular wave which is found to be the more suitable for string instruments.

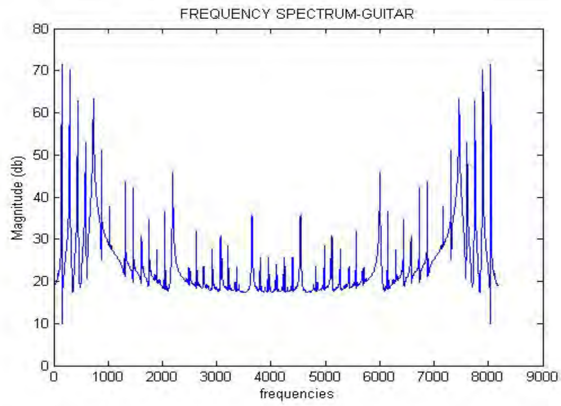

Figure 18: Frequency spectrum of Guitar

Since the periodic wave used in the guitar is the triangular wave which in turn is the sum of various sinusoids, the frequency spectrum of guitar shows various harmonics, which add tonality to the sound. The amplitude envelope of guitar is an exponentially decaying curve. In the guitar it is found that different frequency components decay at different rates. Higher frequency components decay at a faster rate than the lower frequency components. Consequently the higher frequency components have higher energy in the beginning and in the end only the lower frequency components persist creating the ‘booming’ sound of guitar.

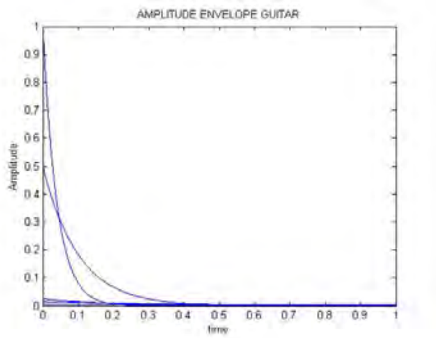

Figure 19: Amplitude Envelope of Guitar

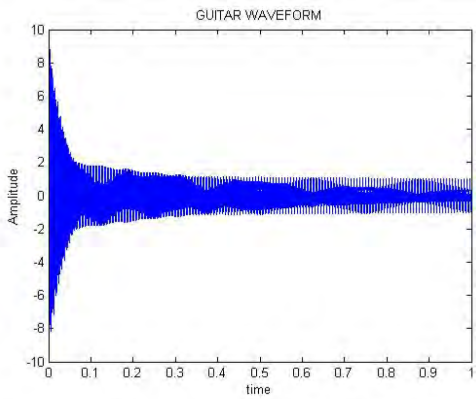

Figure 20: Guitar waveform

IV. AUDIO EFFECTS

Sounds are a part of everyday life. From requisite hearing ability to entertainment in motion pictures and music, sounds have provided the impetus for audio engineers to increase quality of life or level of entertainment through their study and manipulation. We have implemented a digital sound effects processor; specifically, focusing on basic effects such as, delay, delay with feedback, chorus, and flanger. All these tasks are simulated using MATLAB.

Sound effects (or audio effects) are artificially created or enhanced sounds, or sound processes used to emphasize artistic or other content of films, television shows, live performance, animation, video games, music, or other media.

A. Echo:

In its simplest form, a delay effect takes an audio signal and plays it back after a delay time. [12] This “echo device”, as it is otherwise known, produces a single copy of the input, delayed by anywhere from several microseconds to several seconds. A more interesting sound is produced through the use of feedback control, which takes the output of the delay and sends it back to the input, typically after multiplying by a gain less than or equal to one. This enables the sound to be repeated over and over again, becoming more attenuated with each successive loop. Theoretically, feedback can be used to produce a sound which continues forever, but practically, at some point, the sound level will fall below that of the ambient noise in the system, and the sound will no longer be audible.[12]

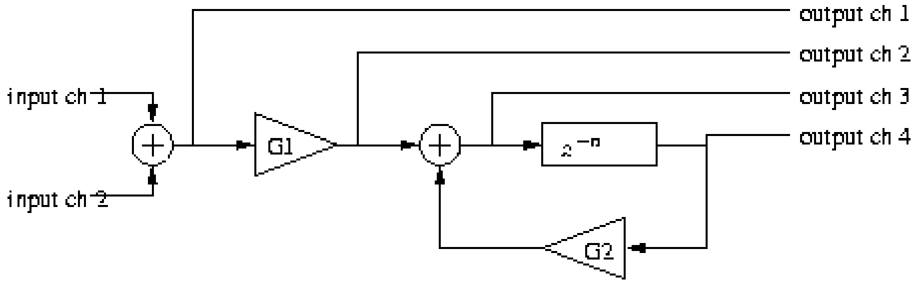

Figure 21: Block Diagram depicting Echo.[13]

B. Reverb:

Reverberation, in psychoacoustics and acoustics, is the persistence of sound after a sound is produced. A reverberation, or reverb, is created when a sound or signal is reflected causing a large number of reflections to build up and then decay as the sound is absorbed by the surfaces of objects in the space which could include furniture and people, and air.[12] This is most noticeable when the sound source stops but the reflections continue, decreasing in amplitude, until they reach zero amplitude. Reverberation is frequency dependent. The length of the decay, or reverberation time, receives special consideration in the architectural design of spaces which need to have specific reverberation times to achieve optimum performance for their intended activity. In comparison to a distinct echo that is a minimum of 50 to 100 ms after the initial sound, reverberation is the occurrence of reflections that arrive in less than approximately 50ms. As time passes, the amplitude of the reflections is reduced until it is reduced to zero. Reverberation is not limited to indoor spaces as it exists in forests and other outdoor environments where reflection exists.

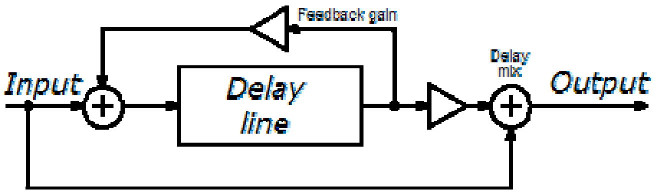

Figure 22: Block Diagram representing Reverb [13]

C. Flanging:

Flanger is much like the chorus, but with a few subtle differences. The flanger also consists of a delay line with varying delay time, but a feedback loop is used as well to generate some distortion in the delayed sound. The common delay for flanger is between 1ms and 10ms which is what gives the input a “whooshing” sound. A low frequency oscillator will be used in this case to control the varying delay line. The same problem will occur here as it did in chorus, so linear interpolation will be used again. Basically, the implementation is similar to chorus, except for a feedback looping and adding to the original signal. [12]

Flanging, or "whooshing" sound is a type of phasing effect. To create it, you add the input signal to its delayed version while varying the delay over time. The following figure shows a block diagram implementation of flanging.

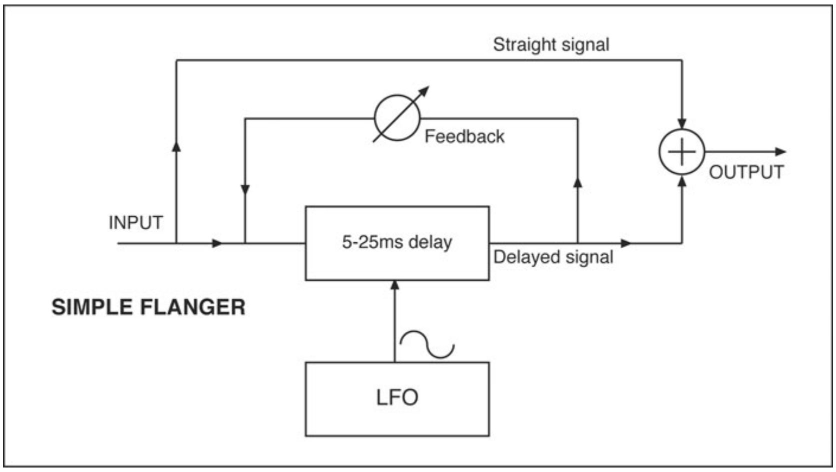

Figure 22: Block Diagram representing Flanging. [13]

The stereo panner adds the effect of the audio signal moving from one speaker to the other. The flanging effect, named after the rim (flange) of audio tape (more on that in a minute), is based on two (or more) parallel audio signals cycling in and out of phase with each other. The sonic result is a whooshing sound that covers ground as diverse as the metallic shimmer of Police-era Andy Summers, the jet swoosh of Barracuda, and the time travelling phase of Hendrix's House Burning Down. Flanging is an audio effect produced by mixing two identical signals together, one signal delayed by a small and gradually changing period, usually smaller than 20 milliseconds. This produces a swept comb filter effect: peaks and notches are produced in the resulting frequency spectrum, related to each other in a linear harmonic series. Varying the time delay causes these to sweep up and down the frequency spectrum. A flanger is an effects unit that creates this effect. In some cases the two signals will become so close that it almost fades away to oblivion called "sucking air." It has also been called the "Darth Vader effect."[12]. Part of the output signal is usually fed back to the input (a "re-circulating delay line"), producing a resonance effect which further enhances the intensity of the peaks and troughs. The phase of the fed-back signal is sometimes inverted, producing another variation on the flanging sound.

D. Chorus:

In music, a chorus effect (sometimes chorusing or chorused effect) occurs when individual sounds with roughly the same timbre and nearly (but never exactly) the same pitch is converged and perceived as one. While similar sounds coming from multiple sources can occur naturally (as in the case of a choir or string orchestra), it can also be simulated using an electronic effects unit or signal processing device.[12]

The chorus effect is especially easy to hear when listening to a choir or string ensemble. A choir has multiple people singing each part (alto, tenor, etc.). A string ensemble has multiple violinists and possibly multiples of other stringed instruments. In spite of the name, most electronic chorus effects do not accurately emulate this acoustic ensemble effect. Instead, they create a constantly moving electronic shimmer.

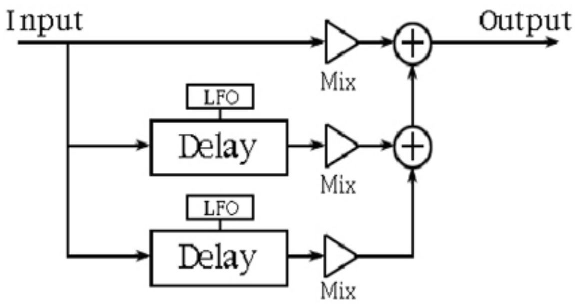

Figure 23: Block Diagram of Chorus effect. [13]

Some instruments produce a chorus effect as part of their design:

- Piano - Each hammer strikes multiple strings tuned to nearly the same pitch (for all notes except the bass notes). Professional piano tuners carefully control the mistuning of each string to add movement without losing clarity.
- 12 string guitars - Six pairs of strings tuned in octaves and unisons create a distinctive complex shimmer.
- Synthesizer - The effect can be achieved by using multiple, slightly detuned oscillators for each note, or by passing all the notes played through a separate electronic chorus circuit.

- Mandolin - 4 pairs of identically tuned strings, as opposed to octaves and unisons on the 12-string guitar.
- Accordion - two or three reed blocks tuned to nearly the same pitch produce a unique and distinctive sound exclusive to the accordion

E. Bass:

Bass describes musical instruments that produce tones in the low-pitched range. They belong to different families of instruments and can cover a wide range of musical roles. Since producing low pitches usually requires a long air column or string, the string and wind bass instruments are usually the largest instruments in their families or instrument classes. Bass describes tones of low frequency or range. In musical compositions, these are the lowest parts of the harmony. [14]

```
%set Parameters for Shelving Filter
% Change these to experiment with filter
G = 4; fcb = 523/2; Q = 50; type = 'Base_Shelf';
[b a] = shelving(G, fcb, Fs, Q, type);
yb = filter(b,a, x)
```

F. Treble:

Treble refers to tones whose frequency or range is at the higher end of human hearing. In music this corresponds to "high notes". The treble clef is often used to notate such notes.[14] Examples of treble sounds are guitar tones, female voice (such as soprano), young male voice, etc. The frequencies generally adjusted by treble controls on audio equipment are above 2 kHz, although vocal and instrumental core tones classed as treble lines are generally lower. Treble sound is the counterpart to bass sound. Bass and treble is implemented using shelving filter. Shelving Filter is used to Boost or cut the low or high frequency bands with a cut-off frequency, Fc and gain G.

```
[B,A] = shelving(G, Fc, Fs, Q, type);
G is the logrithmic gain (in dB)
 FC is the center frequency
 Fs is the sampling rate
 Q adjusts the slope be replacing the sqrt(2) term
 type is a character string defining filter type
 Choices are: ’Base_Shelf’ or ’Treble_Shelf’ [13]
G = 4;  Q = 100;
fct = 1000; type = 'Treble_Shelf';
[b a] = shelving(G, fct, Fs, Q, type);
yt = filter(b,a,x);
```

G. WahWah Effect:

Wah-wah (or wa-wa) is an imitative word (or onomatopoeia) for the sound of altering the resonance of musical notes to extend expressiveness, sounding much like a human voice saying the syllable wah. The wah-wah effect is a spectral glide, a "modification of the vowel quality of a tone" [14]. The wah-wah effect is produced by periodically bringing in and out of play treble frequencies while a note is sustained. The word is derived from the sound of the effect itself in other words, it is onomatopoeic. The method of production varies from one type of instrument to another. On brass instruments, it is usually created by means of a mute, particularly with the Harmon (also called a "wa-wa" mute) or plunger mute. Woodwind instruments may use "false fingerings" to produce the effect. Any electrified instrument may use an auxiliary signal-processing device, usually operated by a pedal. This electronic means is most often thought of in connection with the electric guitar, but is also often used with the electric piano.

```
for n=2:length(x),
   yh(n) = x(n) - yl(n-1) - Q1*yb(n-1);
   yb(n) = F1*yh(n) + yb(n-1);
   yl(n) = F1*yb(n) + yl(n-1);
   F1 = 2*sin((pi*Fc(n))/Fs);
end
%normalise
maxyb = max(abs(yb));
```

yb = yb/maxyb;

H. Tremolo:

Tremolo is a variation in amplitude as produced on organs by tremulants using electronic effects. In guitar amplifiers and effects pedals which rapidly turn the volume of a signal up and down, creating a "shuddering" effect an imitation of the same by strings in which pulsations are taken in the same bow direction.

t=1: length(x); f=5; z=sin (2*pi*f*t); v=z'; y=v.*x

I. Fade in and out:

A fade is a gradual increase or decrease in the level of an audio signal. The term can also be used for film cinematography or theatre lighting, in much the same way. A recorded song may be gradually reduced to silence at its end (fade-out), or may gradually increase from silence at the beginning (fade-in). Fading-out can serve as a recording solution for pieces of music that contain no obvious ending. [14]

```
yin=yin';
step=1/length(yin);
fd=1:-step:(0+step);
fadout=fd.*yin;
y=fadout;
yin=yin';
step=1/length(yin);
fd=0:step:(1-step);
fadin=fd.*yin;
y=fadin;
```

V. GUI DESIGNS

A. The Controller

On the left side of the controller, the user can choose among various instruments including flute, piano, guitar and clarinet. This will open GUIs of instrument chosen. Towards the right there is a master Recorder available where the user can start, play and stop recording as and when needed. Each recording is saved as a .wav file in a default folder at the location of this software. Lastly, the user can choose the effects tab which will open the effects GUI. On the left side of the controller, the user can choose among various instruments including flute, piano, guitar and clarinet. This will open GUIs of instrument chosen. Towards the right there is a master Recorder available where the user can start, play and stop recording as and when needed. Each recording is saved as a .wav file in a default folder at the location of this software. Lastly, the user can choose the effects tab which will open the effects GUI.

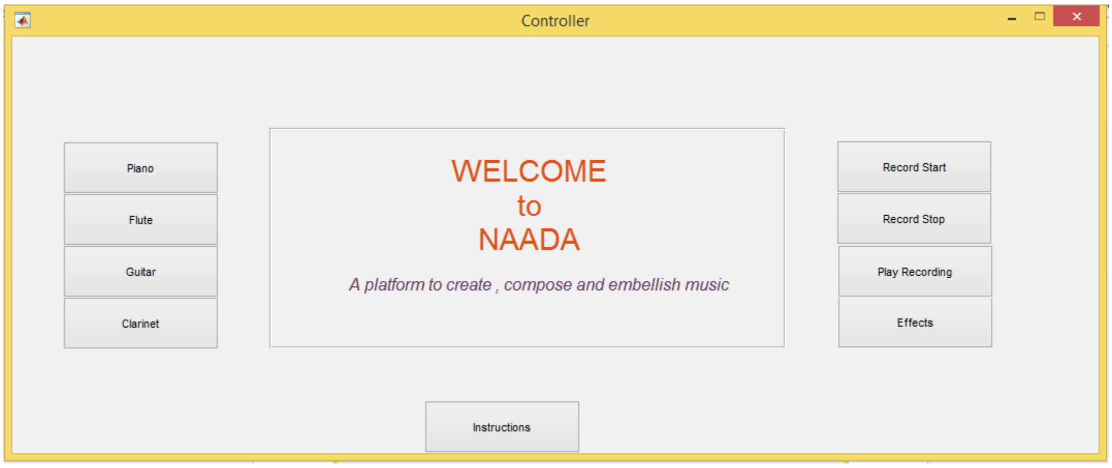

Figure 24: Controller GUI

B. Instrument GUIs.

GUI for various instruments contains keys representing each musical note. The user can click any key using the mouse and can observe a sound related to that key based on the frequency pre-defined as per standard convention of musical notes. The letters denoted in white are musical notes. The frequency is displayed on the top left corresponding to every note played. The user can record, play and save the composition as a .wav file. If the user is not comfortable with the mouse, the user can also use the keyboard using the activate keyboard option.

The letters in red correspond to the key in the keyboard that is to be used. The amplitude plots of the sounds are available on the upper right corner of the screen. The user can also input notes from an excel sheet to play the song in the sounds of the respective instrument, as shown

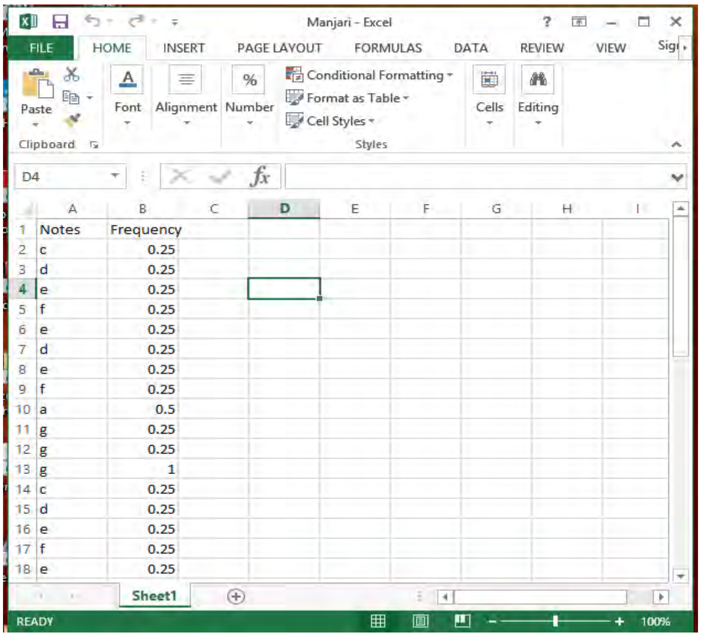

Figure 25: Musical notes input through Exce

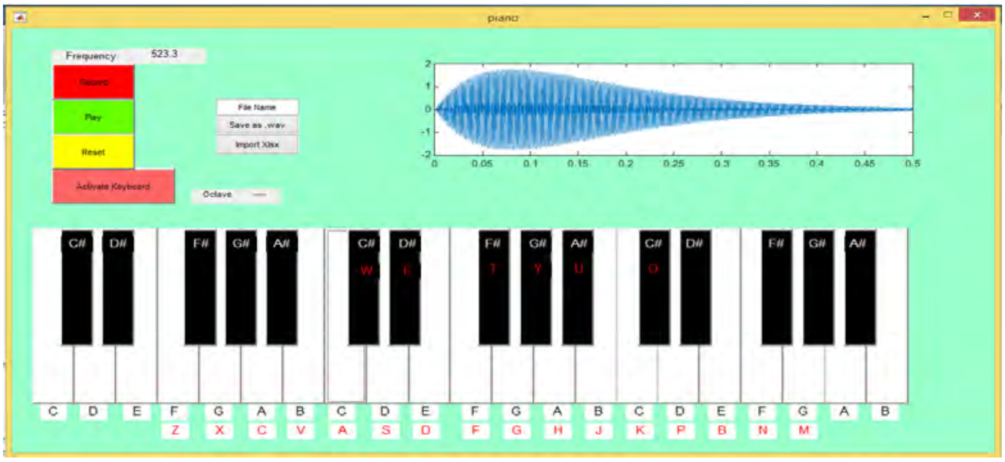

Figure 26: Piano GUI

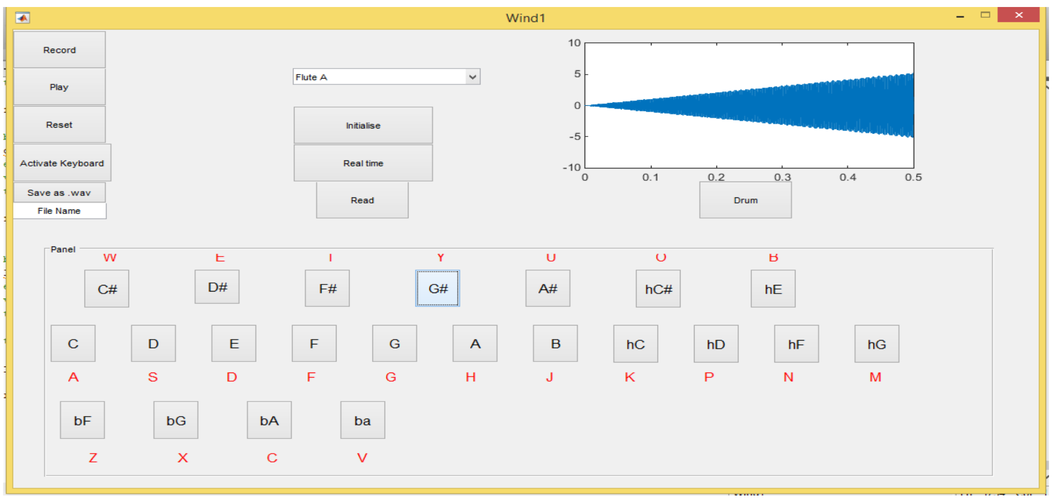

Figure 27: Flute GUI

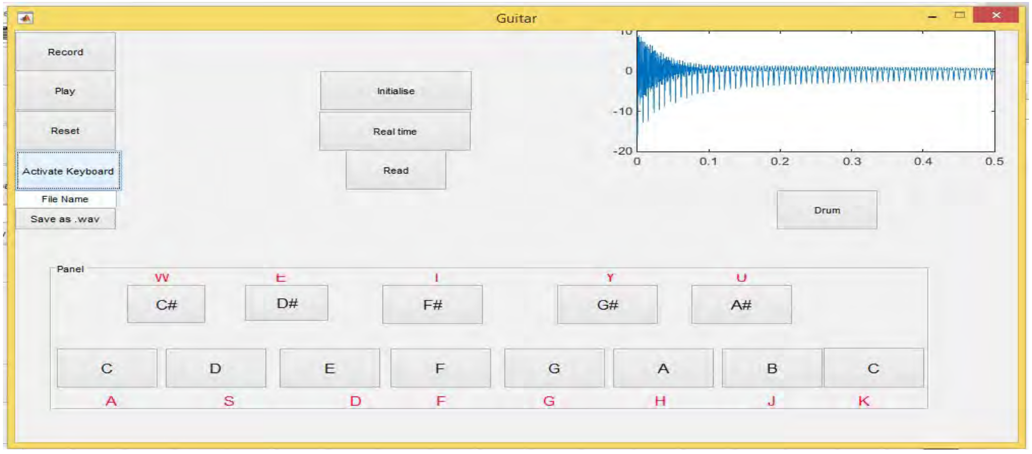

Figure 28: Guitar GUI

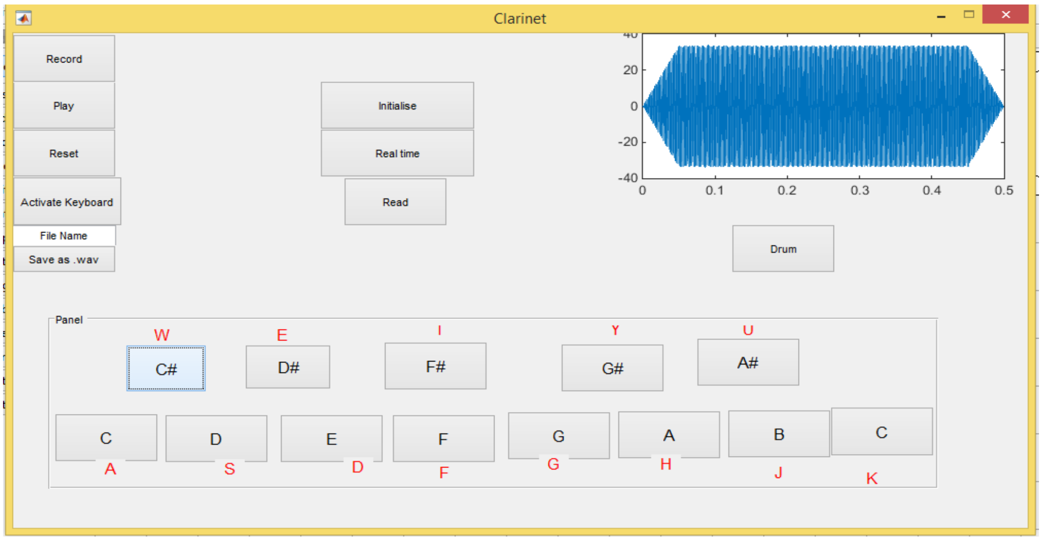

Figure 29: Clarinet GUI

Figure 30: Audio Effects GUI

C. Flow chart of Electronic Music synthesis with audio effects

Flow chart of Electronic Music synthesis and with audio effects is shown in below figure 31.

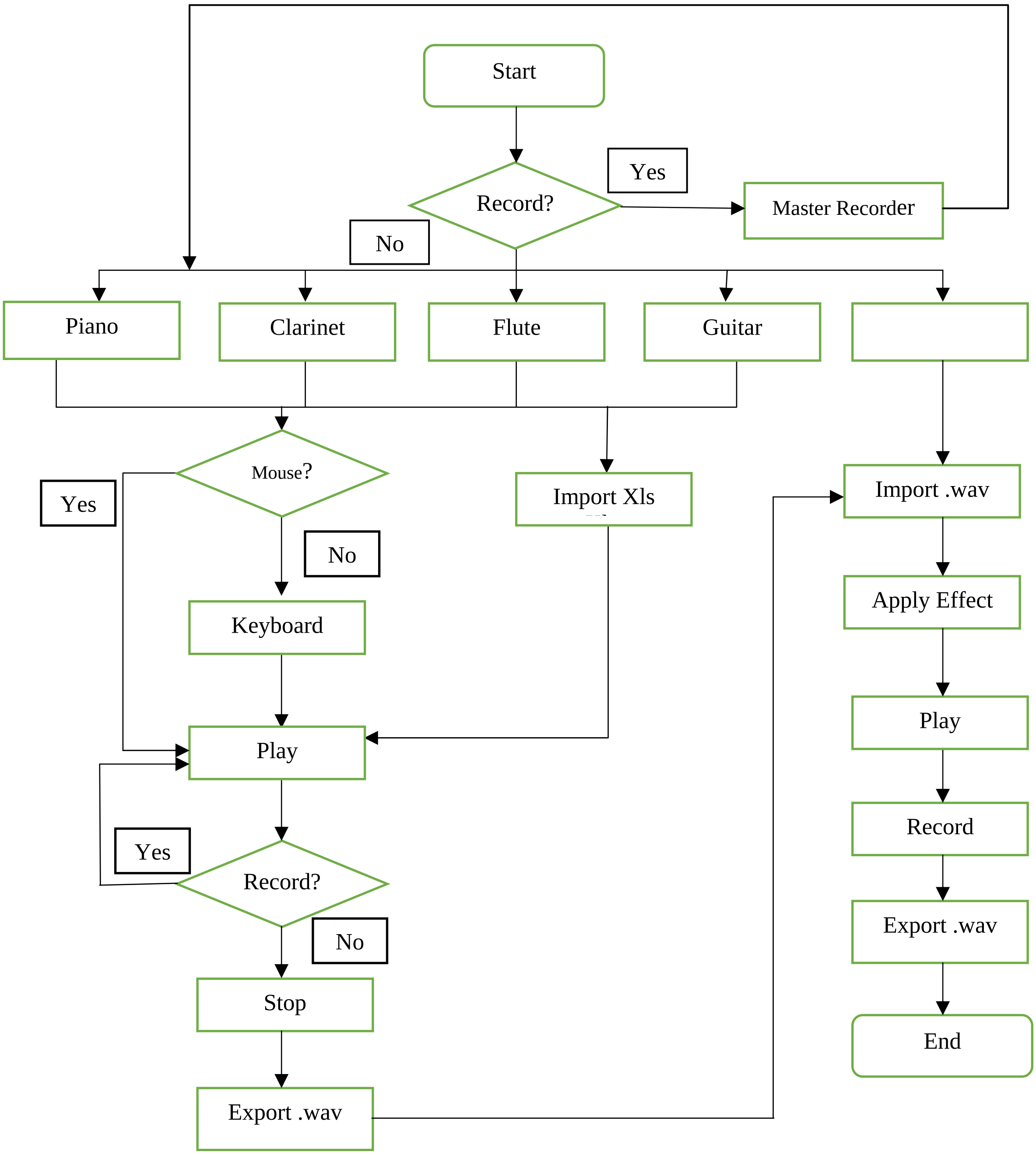

Figure 31 : Flow chart of Electronic Music synthesis and audio effects

VI. CONCLUSION

"Electronic Music synthesis and audio effects" provides an environment to play virtual instruments, compose music and impart audio effects to embellish the generated musical piece and to record and save the musical piece. The sounds of Piano, Flute, Clarinet and Guitar have been synthesized from scratch using physical modelling; no recorded sounds have been used. The most important timbre attributes viz. Frequency spectrum and amplitude envelope have been manipulated to create different timbres. MATLAB environment has been used for this purpose. The most commonly used audio effects viz. Echo, Reverb, Flanging, Chorus, fade-in, fade-out, stereo, wah-wah and tremolo have been implemented using MATLAB. The virtual instruments have been provided in the form of an easy to use MATLAB GUI. Provision has been made for real-time synthesis and recording. The 'read' feature enables the user to directly import the notes of the song and play the same eliminating any human error which might occur during playing.

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