UNIVERSITY of PAISLEY

ELECTRONIC GUITAR HARMONISER

Kamil Kucharski

Supervisor: Mr J. Kinsler

Electronic and Electrical Engineering

5th Year Project

June 1995

DEPARTMENT OF ELECTRICAL AND ELECTRONIC ENGINEERING

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SUMMARY

A method of processing the signal frequency was developed. At first considerations were limited to deterministic signals like sinusoid, rectangle, triangle. Next assumption was that shifting the frequency one octave up for constant frequency of the signal was performed. Once solution to the problem was found signals with changing frequency were considered. Such signals required introduction of a frequency determination method. On successful implementation of the method an additional problem started to appear i.e. changes of amplitude of the signal, so a method for amplitude determination was developed.

At that stage software performed doubling the frequency for signals with changing frequency and amplitude and any type of single wave form separately: sinusoid, triangle, rectangle. However when signal type was changed e.g. from sinusoid to triangle or from sinusoid to triangle and neither frequency nor amplitude change took place, software did not detect signal change and this forced introduction of a method for signal type detection.

At the stage Interrupt Service Routine (ISR) comprised a considerable code and any additional statements in ISR could not be placed in it any more. Hence further analysis of deterministic signals was suspended when this fact was discovered.

Halving in frequency was obtained by a slight modification of doubling of frequency program. The method, however, did introduce some distortion of the signal and a smoothing method for reducing this distortion was developed.

Then efforts were made to find a solution to any arbitrary frequency shift i.e. by semitones. This problem was solved by the means of linear interpolation of samples stored in array.

Finally, the possibility of processing random signals was considered. Another approach was applied and whilst it was not fully tested early indications are that the method is successful.

However the problem of frequency halving was found to be insoluble.

CHAPTER 1 - INTRODUCTION

1.1 COMMENTS ON PREDECESSOR'S RESEARCH AND RESULTS

Background of the project.

One area of particular interest to modern guitarists is that of electronic harmonisers for electric guitars. These harmonisers are used to produce signals which are, in frequency, multiples and sub-multiples of the signal produced by the plucked guitar string. Such pitch shifting is used by solo guitarists who want to enhance their sound by creating the illusion of two or more guitarists playing a harmony. Pitch shifting is an entirely artificial effect and differs from echoing or phasing effects that simulate things which occur in nature. Harmoniser units started appearing in the late 70's, but at the time they were large expensive devices affordable only by professionals.

Currently however, most low- cost multi-effects processors for guitars have some kind of variable pitch shifting facility [Ref.8].

With the availability of a PC-resident DSP interface board, a high level programming language, compiler and linker the DSP chip, many guitar effect programs can be written to be implemented from the PC or from a fully independent package for commercial use.

Continuation of the project. Comments, problems and limitation.

The project reported here was a continuation of a project carried out by John McMillan [Ref.1], whose report was concise and consistent, and contained large sections of code which enabled further research to e carried out.

At the stage at which the shifting frequency program was taken over it performed and could be described by the following features:

(i) shifting the frequency up and down one octave for signals which lasted 0.5 or shorter without discontinuities,

(ii) there were step-like distortions in the signal shifted down in frequency,

(iii) there were limitations to the signal which, when they were not satisfied, resulted in defects of the program:

- there was a delay between signal change at the input and at the output equivalent to 0.5 s since the program detected the signal (frequency, amplitude or shape) changes after the output array was circulated once. In other words there was a dead time of about half a second after the change of the signal had taken place during which the signal at the output was different from the signal at the input.

(iv) the sampling frequency of 41.66 kHz enabled shifting the sinusoidal signal of 8 kHz up one octave and shifting the sinusoidal signal of 15 kHz down one octave, without any loss of the wave form.

(v) there were discontinuities in the output signal when the end of the old samples sequence was faced and the new sequence of samples was the source of processed information. It was so since output array index was traveling along the array with the speed being twice the input array index one.

1.2 OBJECTIVES OF THE PROJECT

1. Develop and improve the method of creating an octave harmony signal, both above and below the input frequency.
2. Implement the developed method in a form of a program used as Interrupt Service Routine with TMS320C30 DSP board.
3. Test and modify the program until the desired effect is achieved with common single waveform signals e.g. sinusoid, triangle, rectangle.
4. Modify the earlier program to harmonize the signal to any semitone step including shifting frequency up an down.
5. Test the program for different wave forms e.g. FM and AM signal.
6. Consider possibilities of applying the method for random signals e.g. noise, speech.

CHAPTER 6 - CONCLUSIONS AND FURTHER RESEARCH

6.1 CONCLUSIONS

Most of the work in pitch shift effect appears to have been performed by the companies that produce the multi-effects units for the electric guitar, without reportage. A search of many DSP journals failed to provide a single report of work in that area. In essence, the work covered here was an original investigation into the workings of a real-time pitch shifter.

It has been shown that frequency doubling and frequency halving for deterministic signals can be obtained in psuedo-real time i.e. with a short delay by the means of basic manipulations performed on samples. It was also shown that frequency shift can be obtained for any arbitrary value for this class of signals.

Frequency doubling was achieved for deterministic and random signals. Frequency halving was only achieved for deterministic signals. The frequency range examined was from 20Hz (the limit of audibility) to 10 kHz (limited by sampling frequency of 44kHz).

Some extra comments on real time processing are necessary.

A delay was assumed in both algorithms. This delay was equivalent to one preceding period in first case and to the period of maximum possible length in second case. No way to avoid this delay was found.

6.2 FURTHER RESEARCH

Although quite promising results have been obtained with frequency doubling for different types of signals, yet the problem of frequency halving for this kinds of signals remains unsolved. This creates an opportunity of further research.

However it is reasonable to presume that for random signals frequency halving in real time cannot be developed since its algorithm assumes time scale expansion and this implies necessity of applying large memory (very long arrays).

There is still some work to be done in spectral analysis of the signal which was processed to find out the influence of the program on this signal. When deterministic signals are considered the applied method yields frequency doubling or halving without any doubt. But for some more complex signals like AM and FM or random signals, this method of processing provides pseudo-doubling or pseudo-halving. Whether this actually is frequency doubling of halving it is possible to find out by the means of spectral analysis. The processed signal definitely comprises double or halve frequency components, but some additional harmonics may also occur and this would necessitate some additional filtering.

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