SAMPLING SYNTHESIS TECHNIQUE APPLIED FOR THE DIGITAL GENERATION OF MUSICAL TONES OF MALAYSIAN FOLK INSTRUMENTS

[1]Ang Yaw Feng & [2]Minni Ang
Music Department
Faculty of Human Ecology
Universiti Putra Malaysia
43400 UPM Serdang
tel. 03-9486101 ext 1738
fax. 03-9161689
1yawfeng@music.upm.edu.my
2minni@music.upm.edu.my

ABSTRACT

Sampling synthesis is a method used for the generation of musical tones. Sampling synthesis permits the production of natural, rich, and time-varying sounds that are useful for composition, live performance as well as sound effects purposes. A random survey of commercial synthesisers available shows that sampled sounds of Malaysian folk instruments such as the rebab, seruling and others have yet to be found in any of these products, be it in the form of software or hardware. On the other hand, Western classical instrument sounds and other varieties of new sounds have long been sampled in those products. To date, no study has been carried out on analysing the waveforms of the sounds produced by Malaysian folk instruments. This paper attempts to present a methodology for the production of a high quality sound bank of selected Malaysian folk instruments by applying the sampling synthesis method. This includes the addition of different effects and modifications. An analysis of waveforms produced by selected Malaysian folk instruments is also presented. Results indicate the presence of many inharmonic partials. In this study, ADSR (Attack, Sustain, Decay and Release) parameters obtained through the waveform analysis are utilised in the production of the digital samples for the purpose of identifying loop points and parameters, which can be modified for compositional purposes. The sound bank thus produced could be stored in the form of software downloadable files or in the form of sound fonts. It is hoped that this sound bank would be useful as a source of musical tones, which can be used in conjunction with a sequencer for the production of MIDI files utilising Malaysian folk instrument timbres.

Keywords: *Sampling Synthesis, Digital Audio, Waveform Analysis, Music Synthesis, Digital Music, Malaysian Folk Instruments*

1.0 INTRODUCTION

Sampling is a process where sound (an analogue signal) is recorded digitally. When sound is recorded into a sampler, it turns the audio waveform into a series of binary numbers or bits (0s and 1s), that can easily be shuffled around and reassembled. This is achieved with an electronic circuit called an analogue-to-digital converter (A/D converter or ADC for short). In contrast, analogue recording is based on voltage recorded as patterns of magnetisation in the oxide particles of recording tape [1]. For instance, in an actual musical instrument, sounds such as individual tones can be recorded, and then played on a keyboard as if you were playing a piano. These individual tones are called "samples" and they can be stored on a CD-ROM or hard disk [2].

The term sampling is derived from established notions of digital samples and sampling rates [3]. "Sampling Rates" in turn, refers to the number of samples that are taken of an analogue signal per second. The more regularly samples are taken, the better the result will be when the sample is played back [4]. This is due to the mechanics of recording devices capable of capturing tiny variations in the sound waves more accurately. As a result, this produces higher fidelity recordings with less distortion [5].

Sampling thus has an edge over multiple waves cycling, in that it applies a longer wavetable containing thousands of individual cycles- several seconds of pre-recorded sounds, permitting the use of pointers within a sample to define internal looping [3]. It creates samples from live and pre-recorded materials. Recorded sound can be stored in disks, or in the internal memory [6]. The sample later can be spliced, copied, reversed, enveloped, cross-faded, looped or sped up, or any combination of the above, in order to change the duration, pitch and timbre [7]. Effects such as reverberation or flanging can also be introduced in the wave-shaping process. As a result, it is usually used to create sonorities and effects that would normally not be possible to create acoustically [8]. In effect, sampling synthesis permits the production of rich, natural, and time-varying sounds useful for composition, live performance and sound effects purposes. It has minimum flexibility since only few transformations are possible at this level [9]. The input signal is always the same, since it is recorded. The input signal is a recorded sound resulting in the absence of the control over life-like qualities of sounds that help enhance the perception of music.

Malaysian folk instruments that have been chosen for this study are as follows: a) String instruments: gambus and rebab (Malay violin); b) Wind instruments: seruling (flute) and serunai; c) Tuned percussion instruments: angklung d) Unpitched instruments: i) Membranophones, or Drums: gedombak, geduk, gendang (bigger drums) and kompang. ii) Idiophones: canang, kesi and tetawak or gong.

This paper attempts to present a methodology for the production of high quality sound banks of selected Malaysian folk instruments by applying the sampling synthesis method. The portion dealing in methodology is divided into five sections: samples recording, waveform analysis, sound editing, adding effects to sound samples and modifying sound samples. The first section deals with pre-recording preparations. In the second section, an analysis of waveforms produced by selected Malaysian folk instruments is presented and the ADSR (Attack, Sustain, Decay and Release) parameters obtained. The third section, sound editing is a method utilised in the reproduction of natural sounds, whilst the fourth section discusses the method applied in the production of sound samples with different timbres, by applying different effects. Finally, new sound samples may be created by modifying different track of sound samples. In the last two sections, the sound samples are processed to change the duration, pitch and timbre.

2.0 LITERATURE REVIEW

Sampling synthesis can be considered a descendant of the tape-based musique concrète[1] [13]. In the 1920s, manipulation of recorded sounds started, culminating in the invention of the magnetic tape-recorder in the 1930s, permitting sequences of recorded sound to be cut and sliced. In the late 1940s,

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[1] Musique concrète refers to the use of microphone-recorded sounds, rather than synthetically generated tones as a pure electronic music.

Pierre Schaeffer founded the Studio de Musique Concrete, in Paris. There, he and Pierre Henry started using tape recorders to record and manipulate concrete sounds. Later, some sampling instruments based on the principle of photoelectric and magnetic tape loops were invented. A few years later, digital electronics devices were invented, allowing sounds to be recorded and stored in digital memory chips. In the 1970s, the first sampling devices were designed, which enriched specific sounds by mixing these sounds with a sampled version of itself, delayed by several milliseconds. 1979 witnessed the appearance of the first commercial keyboard sampler, the Fairlight Computer Music Instrument (CMI), that worked by recording sounds in its digital memory or the process called sampling, costing over US$25,000 [3,7,10]. In 1981, the Emulator (costing about US$9,000) produced by the E-Mu Systems appeared in the market. By 1985, the cost of designing and manufacturing digital hardware had declined to a more attainable level, and this made possible the introduction of the Mirage by Ensoniq. The Mirage was just the first series of a series of increasingly sophisticated, relatively inexpensive, sampling devices manufactured by Ensoniq, Roland, Akai, Casio, E-Mu, Kurzweil, and others. Today, all digital audio technology is based on sampling technology. From sampling keyboards, drum machines, DAT recorders, DJ mixers, digital audio workstations (DAW), multi track digital multi-tracks (MDM) and more [1]. It is now widely used in commercial electronic instruments and multimedia systems too [9]. Moreover, sampling is the most easily available synthesis technique in commercial software [11, 12].

3.0 METHODOLOGY

The steps involved in the production of a high quality sound bank of selected Malaysian folk instruments by applying the sampling synthesis method are described below. The methodology is made up of five main steps: the samples recording, the waveform analysis, the sound editing, adds effects to sound samples and modify sound samples. (Figure 1).

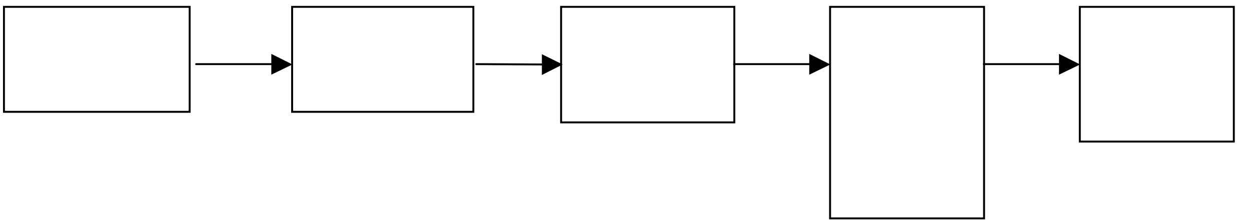

Figure 1. Summary of the methodology

3.1 Samples Recording

Before recording samples, the following criteria are fulfilled in order to get a good sample: a good microphone, a quiet environment (to eliminate extraneous noise), a good technique to get exactly the sound required, a good instrument, an accomplished player who can accurately tune the instrument to be recorded to bring out all the shimmering overtones, and sufficient time to get it right [14]. The sound of the Malaysian folk instruments concerned is recorded using Digital Audio Tape (DAT). The setting of both dynamic microphones (having the bi-polar pick-up pattern) is X-Y coincident miking, where the capsules of the two microphones are placed together and aimed in directions that are between 90^0 and 135^0 apart. Before recording, the LIMITER of the DAT recorder is set to ON, to avoid distortion while recording. For all the pitched instruments, only every third or fourth note of the instrument is recorded, in order to reduce the large amount of memory required to store the samples. The recorded sound is later transferred to a computer using a connector, and saved in the memory of the computer. The production of the complete range of notes is then implemented by applying the pitch shift effect to the sampled note.

3.2 Waveform Analysis

The second part of the research concentrates on briefly analysing the waveform of the sound sampled. From the shape of the waveform, the frequency components (partials) that appear in specific instruments, and the spectrum envelope, ADSR (Attack, Decay, Sustain and Release) are identified. These parameters show the relationship of the instrument's waveform volume in relation to changes with time.

3.3 Sound Editing

The third section involves editing the sound in order to produce a track of natural sampled sound. The saved sound samples are converted into the hardware sampler format utilised, and loaded for further processing. Before editing the sample, noise reduction is applied to the entire waveform in order to filter out any noise that occured during the recording process. The waveform is trimmed to lop off unwanted parts at the start and end points, and normalised to optimise the sample volume. To further reduce memory requirements, resampling and looping are applied to the sampled sound. The sound sampled is resampled at lower sampling rates. This is especially for the instrument sounds that do not have a high harmonic content: for instance, the rebana can be resampled from 44100Hz to 22050Hz or lower sampling rates, as it consists of lower pitched sounds. Next, specific instruments, such as string and wind instruments, are looped to extend sampled sound duration. This is done by finding the best loop point, either automatically or manually, through zooming in to find a smooth transition by selecting a segment whose beginning and ending match closely in amplitude, pitch and tone colour. A smooth transition repeated loop could be achieved by applying crossfade loops. Time stretching is then applied to change the length of a sample, shortening it or lengthening it without changing its pitch. This step ensures that all samples are of uniform duration. Finally, the complete range of notes is obtained by shifting the pitch from stored samples, using the pitch shifting parameter, without changing its duration.

3.4 Adds Effects To Sound Samples

In the fourth section, psycho-acoustic and special effects such as chorusing, flanging, phasing, panning, echo and reverberation are added to the sampled sound in order to produce new effects with different timbres. This can be achieved using the multi-effects channels in the hardware sampler, which also provides distortion, ring modulation and equalisation (EQ), other than a choice of modulation effects from the original sound that was mentioned earlier. Adding ring modulation can produce a variety of different effects such as tremolo, robotic, metallic effects and others, while adding distortion to drums can produce a more interesting sound. By adding equalisation (EQ) to a sound, the frequency range and gain of different frequencies may be set to produce different effects. Some samplers also offer rotary speaker effects that work by spinning a speaker round a motor, giving a very pleasing effect. The slow speed gives a smooth undulating effect, while the fast setting gave a kind of tremolo-cum-vibrato effect.

Software may also be used to apply effects such as:

- Decay effects: chorus, decay, echo, echo chamber, flanger, multiple decay, reverb and sweeping phaser.

- Filter: DTWF/North filter, FFT filter, graphic equaliser, parametric filter, quick filter and scientific filter.
- Special effects: convolution, distortion

3.5 Modify Sound Samples

At the final section, the different tracks of sound samples are joined, reversed, modified and merged to create new sounds. For instance, a particular spectral envelope track, such as that from the rebab, may be applied to one another different track, such as the seruling, to achieve this purpose. This produces new and innovative sounds.

3.6 Software Sampling Synthesis

The sound bank produced may be imported and stored in the form of software downloadable files or in the form of sound fonts using software sampling synthesis. The new instruments can then be used as a sound source to play back a midi file. The sampled voices are created by taking the sampled sounds and using them like standard MIDI tones. It is thus practical for the computer musician. The advantages of these instruments is that their sound quality should not be affected by the sound card being used. Furthermore, certain effects and filtering that are provided in the software used may be applied to the instruments, using the parameters provided. Here, all the pitches of the instruments will be automatically shifted and be able to play back a range of notes.

4.0 RESULTS

This section only presents an example of results obtained, due to space constraints: an example from one of the Malaysian folk instruments: *Gong Ibu.* Figure 2 shows the waveform of the Gong Ibu. From Table 1, the total duration of the waveform is 14439 ms, with only 12 ms of attack time and zero decay time, characteristic of the percussion instruments. *Gong Ibu* has a very long sustain time and an even longer release time, with 306 ms for sustain and 9121 ms for release. Figure 3 shows the ADSR Envelope of the *Gong Ibu*.

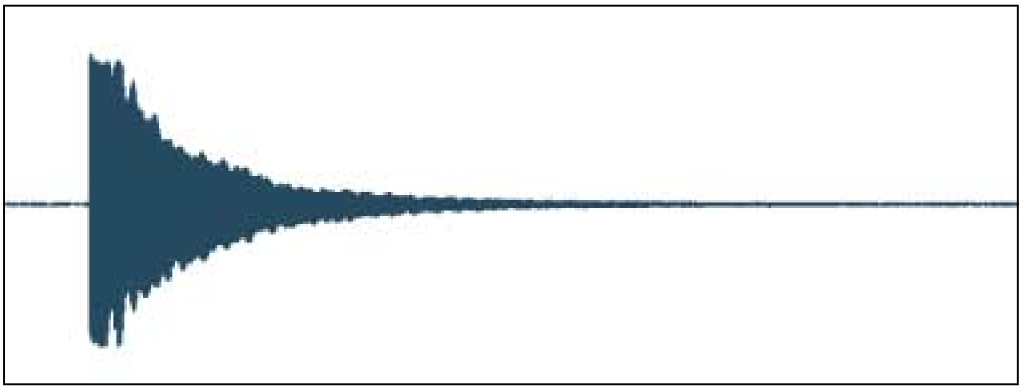

Table 1. ADSR envelope of *Gong Ibu*

No	*Instruments*	*A (t/ms)*	*D (t/ms)*	*S (t/ms)*	*R (t/ms)*	*Total (t/ms)*
1	Gong ibu	12	0	306	9121	14439

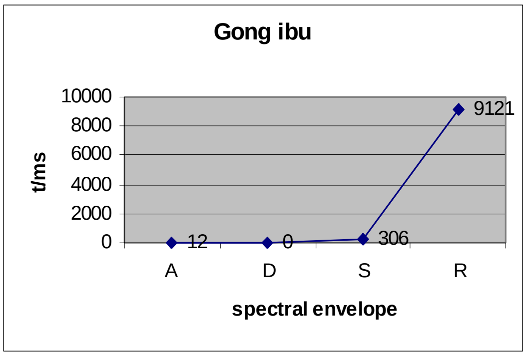

Figure 3. Graph of *Gong Ibu* ADSR Envelope

Figure 4 shows the graph of the frequency components that occur in the *Gong Ibu* tone, while table 2 shows the frequency components and their amplitudes. The results indicate the presence of many inharmonic partials in the *Gong Ibu* tone due to its long period of resonance.

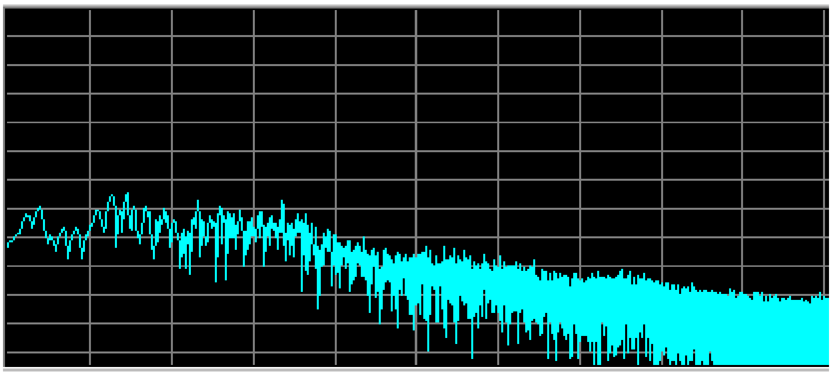

Table 2. Frequency components and the amplitudes of *Gong Ibu*

No	*Frequency(Hz)*	*Amplitude(dB)*
1	143.5	130.27
2	287.3	129.90
3	353.2	121.39
4	437.6	100.16
5	575.0	97.46
6	722.9	98.17
7	875.8	93.62
8	929.1	79.69
9	1000	77.55
10	1076	75.10
11	1150	86.17
12	1211	75.18
13	1295	82.86
14	1304	85.64
15	1363	78.44
16	1446	62.35

The sample of the recorded sound of the *Gong Ibu*, having thus been analysed, is further processed according to the steps outlined in the methodology section, to produce an electronic instrument voice, which may be loaded and used for playback.

5.0 DISCUSSION

The continual efforts to reduce the memory requirement of the stored samples, such as only sampling the third or fourth note of each musical instrument range, resampling to a lower sampling rate where possible, and looping sample sections to obtain the effect of sustaining a particular note, were carried out due to two reasons. The first of these is to reduce the disk storage space required for the samples, and for any voices that may be created utilising these samples. Despite the advances in digital storage technologies, many sampling devices such as keyboard samplers still use the floppy disk as their primary means of loading new sample data. The quest for minimising storage is thus aimed at making the sample bank developed accessible to as many potential users as possible. While it may not be possible to store the entire bank on one floppy disk, at least it should be possible to store a small number of samples on a single disk, allowing the user to transfer required samples to their devices through this means if necessary.

The second reason for the minimisation of memory requirements is to ensure that the bandwidth required for the transmission of data from the sample bank to the digital-to-analogue converter on the playback device is kept at a minimum. This is to avoid the possibility of introducing a latency time factor, where a perceptible gap is heard between the action of triggering the playback of a sampled voice and the actual output of that voice, caused by the time taken to transfer too large amounts of data. Optimising the memory requirement during all stages of the sample bank creation process helps to avoid this problem.

6.0 CONCLUSION

In this paper, a methodology for the production of a high quality sound bank of selected Malaysian folk instruments, by applying the sampling synthesis method, has been presented, including a detailed presentation of the approach to the analysis of waveforms for the purpose of identifying salient features in recorded samples.

The sample bank of Malaysian folk instruments may be used as a tone source for commercial applications. It can be used as an internal tone bank on the commercial products such as keyboards, synthesisers, and tone generators. In addition, the sample banks can be produced as a compact disk and marketed. It can also be produced as an add-on ROM chip or add-on voice-bank card.

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