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The Sound Synthesis and the Establishment of Computer Music Environment

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Abstract

This paper depicts the fundamental sound synthesis and the education for the computer music composition theory and methodology, to establish the education environment for the computer music in the digital art education. The sound synthesis methods, including additive synthesis, subtractive synthesis, amplitude modulation synthesis, frequency modulation, are all important for the sound synthesis in the field of computer music. This paper uses the brief mathematic expressions and the related software to do the analysis and implementation, to achieve the goal to help the people who attempt to compose pieces via computer music with clear and effective ways. The fundamental rules from acoustics and physics can be also applied to tightly integrate both music and technology, to establish the environment for the compute music composition in the field of digital art education.

Keywords: computer music education environment, acoustics, sound synthesis.

Introduction

Computer music utilizes the methodology of music technology to perform the analysis, synthesis, transform, and integration of sound, to attain the goals in music composition and education. The mainstream in music composition is gradually oriented into the trending of electronicalization, digitalization, and internationalization. For instance, there are numerous departments and institutes related to electronic or computer music under computer science or art colleges, all over the countries in Europe and USA. Therefore the electronic or computer music is the important means and direction to develop the contemporary music, and the pieces composed by the composers including Stockhausen in Germany, Pierres Schaeffer and Pierre Boulez in French, Iannis Xenakis in Greek, and Edward Varese and John Cage in united states accordingly. Due to the huge amount of artists have been devoted themselves to electronic and digital art, the twenty-first century is the new era of digital art technology. The excellent features including transformation, muti-track juxtaposition, and stochastic and algorithmic composition provide the variety and convenience for the electronic or computer music, to not only inject a motive power for the new music, but also integrate the sound synthesis and transformation techniques, to create the brand new compositional style for the digital sound art and the art education methodology.

The Relation among Acoustics, Physics and Computer Music

(1) The Difference between Audio and Music

Many people treat audio and music as the same thing. Actually there are s ignificant differences including the follows:

(i) Audio is physical phenomenon, and its parameters only involve three things: amplitude, frequency, and time. The correspondent time-domain relation for the amplitude versus time of the sound wave signal is depicted in Figure 1, with imported by the audio processing software Audition, and the horizontal axis represents time, and the vertical axis represents amplitude. The frequency-domain spectrum for digital sound wave is shown in Figure 2 to express the relation between amplitude and frequency, and the horizontal axis is frequency, while the vertical axis is amplitude.

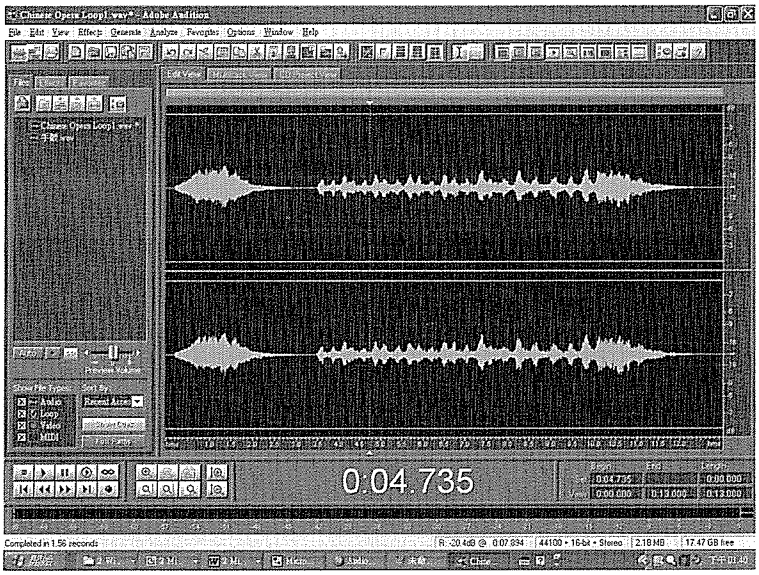


Figure 1. The Sound Wave in Time Domain

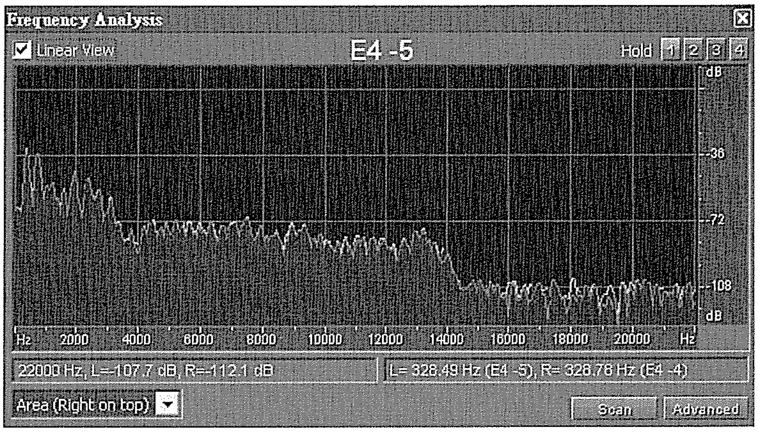


Figure 2. The Spectrum of a Sound Wave

(ii) The Music is psychological, including many parameters, such as: pitch, duration, rhythm, texture, articulation, dynamics, harmony, counterpoint, and tempo, etc.

Generally speaking, the computer music composers use the psychological music parameter settings to compose music, and the computer music programs turn these musical parameters from composers' settings into the physical sound parameters with signal process to complete their pieces.

(2) Harmonic Series and Harmonics

In the development process of the computer music, acoustics plays an extremely important role, especially the "harmonics" in the field of music actually is the "harmonics" in the field of engineering, and then the harmonics distribution basically determines the timbre change. If the two areas of music and physics can be thoroughly investigated and integrated, then the sound signal can be processed via the engineering methodology. Figure 3 shows the lowest note "C" as the fundamental note, to generate the harmonic series, with the fundamental frequency f, the second partial frequency is 2f, and up to the 16th partial frequency is 16f, etc.

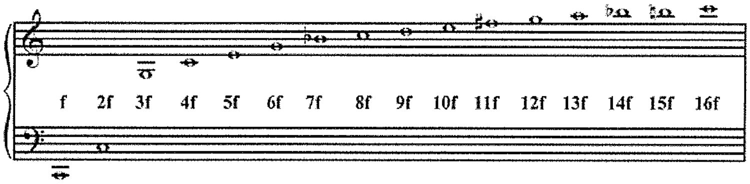


Figure 3. The Harmonic Series with Relations in Frequency Domain

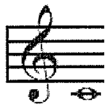
Except the notes in the frequencies of f, 2f, 4f, 8f, and 16f from the above figure, that is, excluding all of the octaves of the fundamental C note, the remaining harmonic partials can not be played with the piano keys accurately. It is due to the twelve-tone equal tempered tuning system, which is out of the just intonation tuning system for the harmonic series.

The lowest pitch C as the fundamental tone in the above figure is:

The mathematic expression for the harmonic series is based on the sound wave signal of the fundamental note:

$$y_1(t) = A_1(t) \bullet sin(2\pi ft + \phi_1)$$

If the fourth partial is middle C:



the signal can be expressed as follows:

$$y_4(t) = A_4(t) \bullet \sin(2\pi f_4 t + \phi_4)$$

Therefore complete expression of the sound wave with fundamental note f to include the infinite harmonic series can be defined as the follows:

$$y(t) = \sum_{n-1}^{\infty} A_n(t) \bullet \sin(2\pi f_n t + \phi_n)$$

where f_n is the frequency of the nth partial harmonics, i. e. :

$$f_n = nf$$

When n=1, $f1=f$ which is the frequency of the fundamental note. ϕ_n is the phase angle of the nth partial harmonics. $A_n(t)$ is the amplitude of the nth partial harmonics.

With the superposition of the partials of the harmonic series, the sound wave can be built into a complex wave, to perform all kind of audio transformations via signal process

(3) Doppler Effect

Doppler Effect utilizes the radial velocity change between the sound source and the listener, to generate the frequency change, and then the pitch change can be obtained from the computer music.

The Doppler effect can be expressed as follows:

$$f_L = f_S \left(\frac{a+V_L}{a+V_S}\right)$$

where f_s is the transmitted frequency of the sound source, f_L is the frequency for the listener, a is the sound speed, and usually is 340 meters / sec in the normal room temperature, V_s is the motion speed of the sound source, and V_L is the motion speed of the listener.

The listener can detect the sound frequency, f_L which varies with the change of V_L and, V_s therefore if the listener and sound source is approaching into the radial direction, then the listener will sense the rising change in frequency. On the contrary, if the listener and the sound source is departing, then the listener will sense the descending frequency change, therefore the frequency change for the sound pitch can be generated by the Doppler effect.

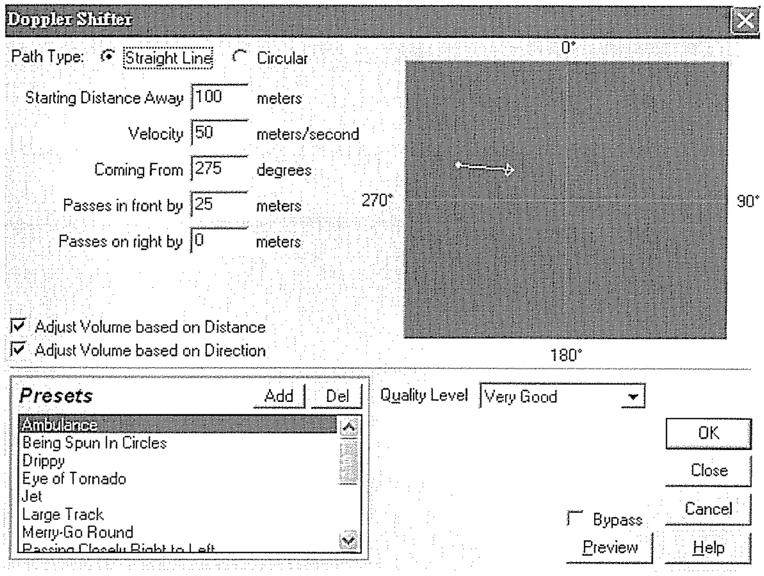


Figure 4. The Settings for the Doppler Effect of a Sound Wave

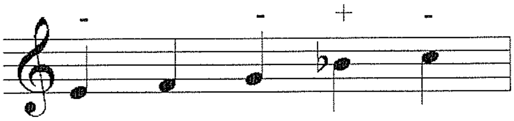
(4) The Beating Effect

The beating effect is to use two slight detuned pitches to sound together, and the resultant amplitude and frequency will change rapidly and periodically with dazzle. For instance the bronze instruments in Balinese Gamelan usually are intentionally detuned with slight different pitches for a pair of them, to vibrate the sound wave with shimmery, to generate the beating effect.

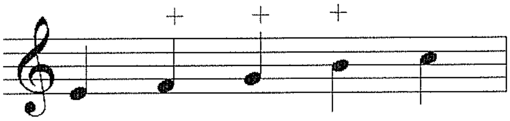
The beating effect in Gamelan music can be analyzed via the computer program, such as the mathematic analysis program Matlab to simulate the result. For example, the tuning system in Sulaan and Gianyar will slightly be detuned for a pair of Selisir Gong into the tuning system of the following scales to generate the "beating" effect. The "+" and "–" mnemonics in the score represent the slight higher and lower pitches than the notated notes, respectively.

There are two sound waves with closed frequencies, one is 20 Hz, and the other one is 17Hz:

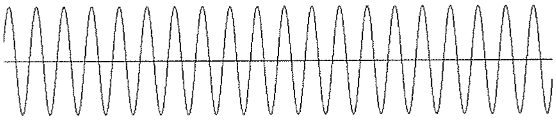
Sulaan



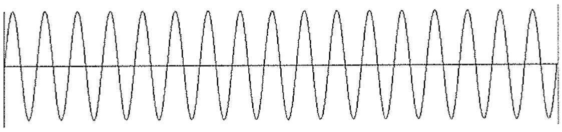
Gianyar



20Hz



17Hz



then the one second long "beating" effect can be represented as the following diagram:

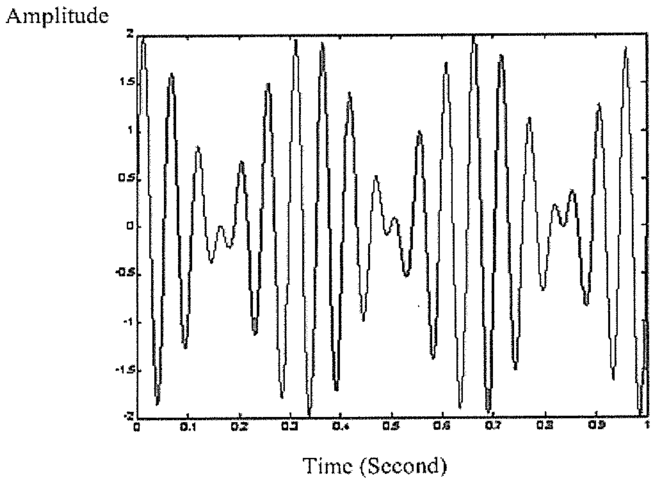


Figure 5. The "Beating" Effect for a Sound Wave

The Applications for the Computer Music in Sound Synthesis Methodology

(1) The Additive Sound Synthesis

The additive sound synthesis superposes the frequency f = f1 of the fundamental note and the frequencies of the partials of the harmonic series. Figure 6 shows the sinusoidal waveform of the fundamental note frequency.

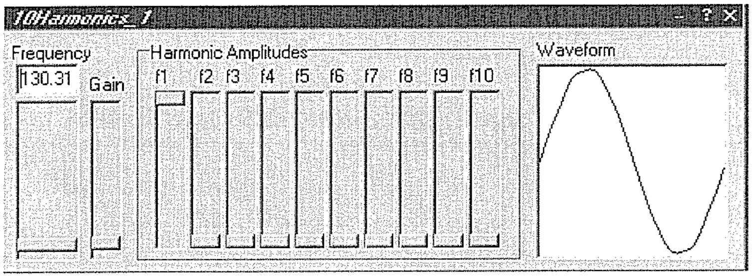


Figure 6. The Sinusoidal Waveform of the Fundamental Note Frequency

The following figure shows "10 Harmonics" object in the audio process software AudioMulch to set the fundamental frequency f = f1 = 130.31Hz, then the 2nd partial frequency is f2 = 2f, and the 3rd partial frequency is f3 = 3f, and so on. Using the additive sound synthesis, we can also use the method "only set the amplitudes for all of the odd number partials, and the amplitudes are inversely proportional to their odd number" to perform the square wave sound synthesis. For instance, f3 = (1/3)f, f5 = (1/5), etc.

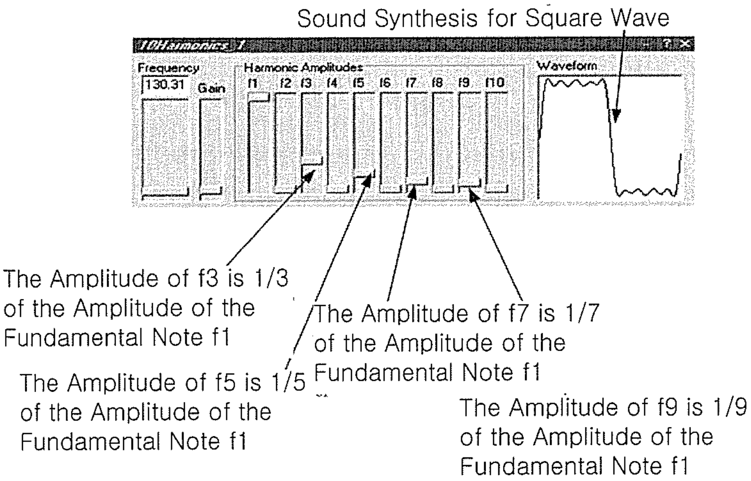


Figure 7. The Sound Synthesis for Square Wave

(2) The Subtractive Sound Synthesis

The filter design can be applied to remove some specific band of the audio signal, to attain the "subtractive sound synthesis". For example we can generate a period of noise, no matter the white noise or the other kind of noise with variant distributions, and the "notch filter" can be used to reduce or remove some certain frequency band of the sound signal. Figure 8 shows the attenuation or even elimination of the frequency band 2K~5.6K Hz for the audio signal by the filter design. This is the basis of the so-called subtractive sound synthesis.

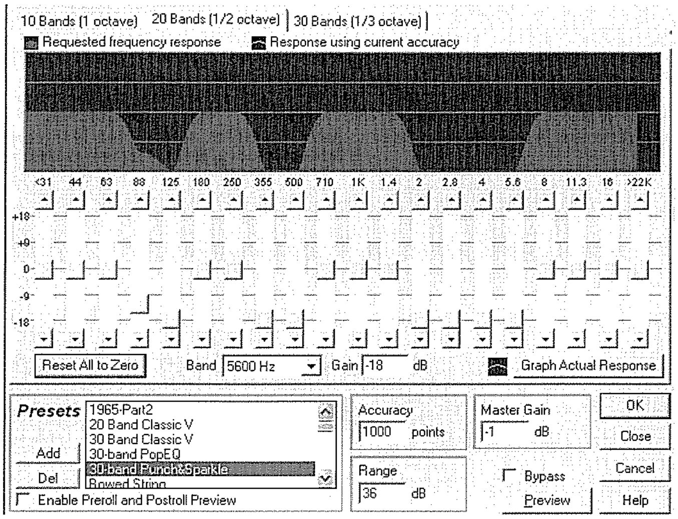


Figure 8. The Subtractive Sound Synthesis

(3) The Sound Synthesis of Amplitude Modulation

The Amplitude Modulation (AM) utilizes the modulator signal to vary the amplitude of the carrier signal. For instance the low frequency oscillator (LFO) can be used to generate the modulator signal, and the voltage controlled oscillator (VCO) can be used to generate the carrier signal, and the resultant signal can be expressed as follows:

$$y(t)=\sin(2\pi f_m t)\bullet\sin(2\pi f_c t)$$

where f_m and f_c represents the frequencies of the modulator signal and the carrier respectively.

with the calculation of the trigonometric function, the audio signal after modulation can be obtained as:

$$y(t)=\frac{1}{2}\ [cos(2\pi\,[f_c+f_m]-cos2\pi\,[f_c-f_m]]$$

Two frequency components f_c+f_m and f_c+f_m can be obtained from the spectrum.

The "Ring Modulation" can be generated via the computer music program Max/MSP, by using two "cycle~" objects' to multiply their generated sine waves. The following figure shows the setting for f_c= 800 Hz and f_m = 0.2 Hz.

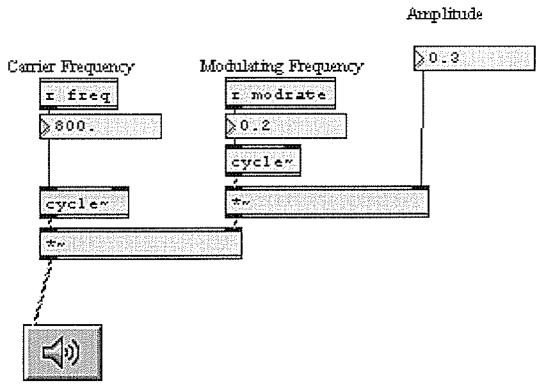


Figure 9. The AM Sound Synthesis using Max/MSP

Now we can use the audio process software Audition to record the AM sound wave generated by Max/MSP, as shown in the following figure:

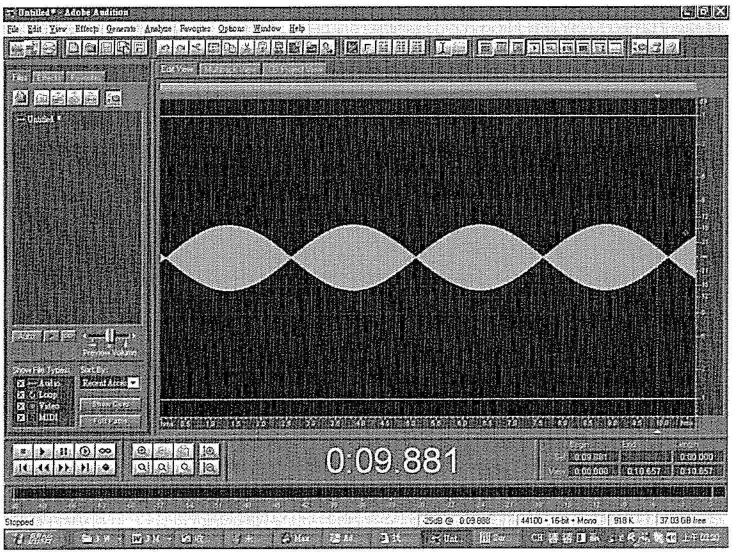


Figure 10. The Waveform for the Ring Modulation of AM

The audible 800 Hz audio is the carrier. However the signal with 0.2 Hz frequency imposed on the carrier amplitude, as shown in the above figure, is the modulation signal. This kind of AM sound synthesis is used as tremolo applied for the computer music field.

(4) The Frequency Modulation Sound Synthesis

The frequency modulation (FM) applied in the field of sound synthesis was derived by John Chowning at Stanford University in 1973. Similar to the AM sound synthesis, FM also utilizes the modulation to alter the frequency characteristics of the carrier. (Note: AM alters the amplitude characteristics of the carrier) The signal expression is as follows:

$$y(t) = \sin[2\pi f_c t + \mathrm{I} \sin(2\pi f_m t)]$$

where I means modulation index, used to control the distortion of the carrier.

FM signal can be calculated into the following equation:

$$y(t) = \sum_{n=0} J_n\ (I)\ [\sin(2\pi[f_c + nf_m]\,t) + \sin(2\pi[f_c - nf_m]\,t)]$$

where $J_n(I)$ is the coefficients of the Bes∑sel function, to represent the amplitudes of the frequency components.

The frequency components $f_c \pm f_m$, $f_c \pm 2f_m$, $f_c \pm 3f_m$, and so on, can be obtained from the spectrum.

FM sound synthesis can be implemented by the computer music program Max/MSP, using two "cycle~" objects to generate the sine waves, with the formula of FM, with the setting of f_c=60 Hz, f_m=3 Hz, and I=15, as shown in the following figure:

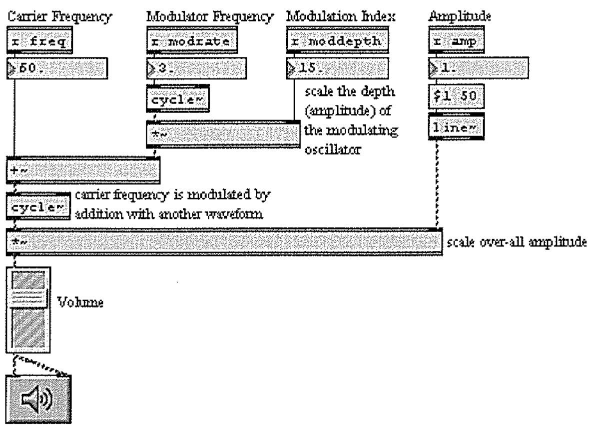


Figure 11. The FM Sound Synthesis Using Max/MSP

The FM sound wave can be recorded by the audio process software Audition, and the time-domain waveform is as the following figure:

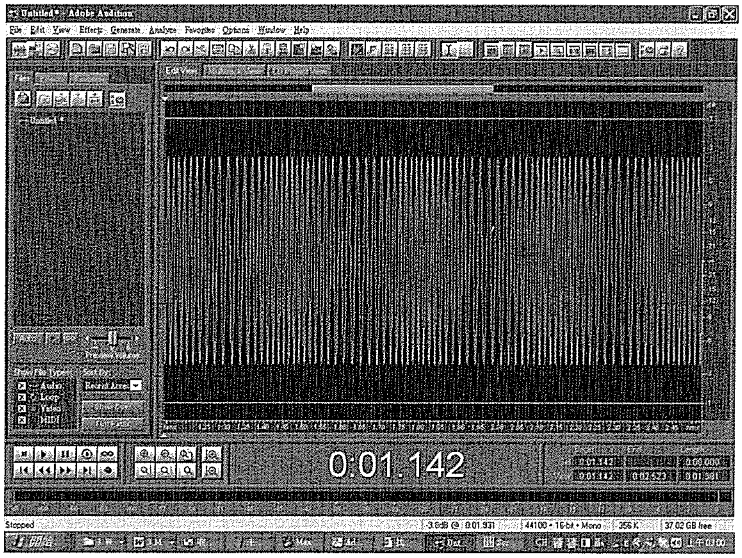


Figure 12. The Waveform of the FM Sound Synthesis

Based on the above figure we can perceive that there is no amplitude change in the FM sound wave, however the frequency alteration can be observed by the periodic density change.

The Discussion for Methodology of the Computer Music Education and Composition

(1) The Direct Sound Synthesis

Sound waves can be generated from the superposition of all kind of oscillators. For instance the sine wave, square wave, saw-tooth wave, and the triangular wave can be obtained by the voltage controlled oscillator (VCO), and the modulation signal can be generated by the low frequency oscillator (LFO), then the amplitude envelope can be designed as the amplitude control, to complete the sound synthesis with frequency modulation, amplitude modulation, and any other kind of the sound synthesis.

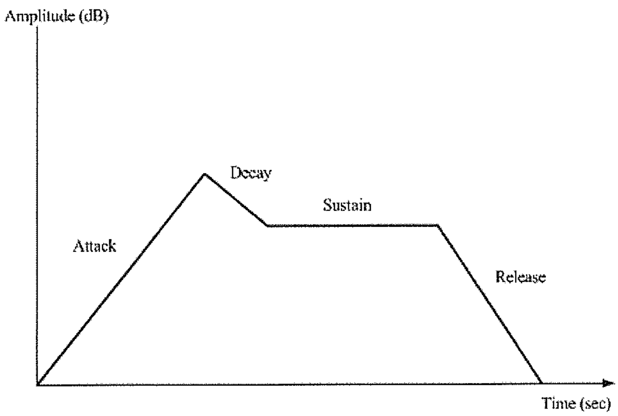


Figure 13. ADSR Amplitude Envelope

The amplitude envelope determines the articulation of the timbre of the instrument. For example the percussive sound owns much shorter attack and decay time, than the other orchestration instrumental sound has the longer sustain and release period of time.

(2) The Indirect Sound Synthesis

The sound wave samples can be imported into the audio process software with every kind of transform technique to alter the characteristics of the sound wave. There are many sound transform methods in the audio process software Audition, including Time/Pitch, Reverberation, Filters, etc. The following figure shows the selected sound sample with the Time/Pitch transform technique.

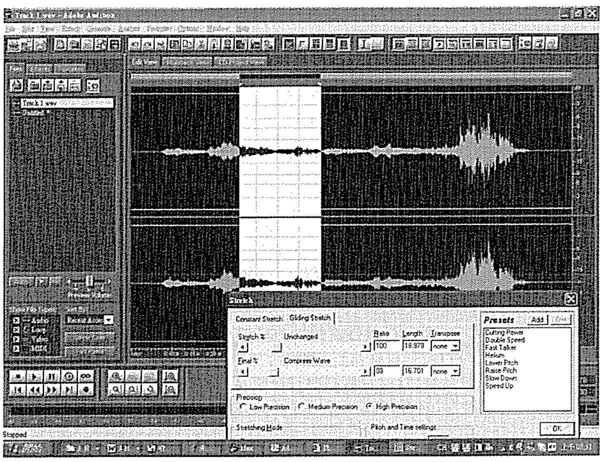


Figure 14. The Sound Stretch Transformation

(3) The Role that Sonogram Plays in the Computer Music Composition and Education

Sonogram or so-called Spectrogram uses Short-Time Fourier Transform (STFT) to transform the audio signal from the time domain into the dynamic spectrum in frequency domain continuously, as the expression in the following formula.

$$STFT(\tau,w) = \int_{-\infty}^{\infty} x(t) \bullet w(t-\tau)\, e^{-jwt} dt$$

The sonogram in the computer music program can be used to import the sound sample to express the relation among time, amplitude, and frequency. It is very useful for the fields in composition, analysis, and education of the computer music. The Sonogram especially provides an excellent analysis methodology for the details of the computer music composition which can be expressed in the traditional score.

The digital form of the STFT can be expressed as the follows:

$$STFT(m,w) = \sum_{-\infty}^{\infty} x[n]\, w[n-m]\, e^{-jwn}$$

Its amplitude of the sonogram is:

$$Mag = |STFT(\tau,w)|^2$$

The following diagram shows the 3D surface of the sonogram of the "Pasibutbut" song from the Taiwanese aboriginal Bunun people's octophonic chorus. The frequency distribution of the bass tone is broad and successive, and the frequency of the bass tone of the leading singer shows the trending in gradual ascending. The distribution of the harmonics is significant and contiguous.

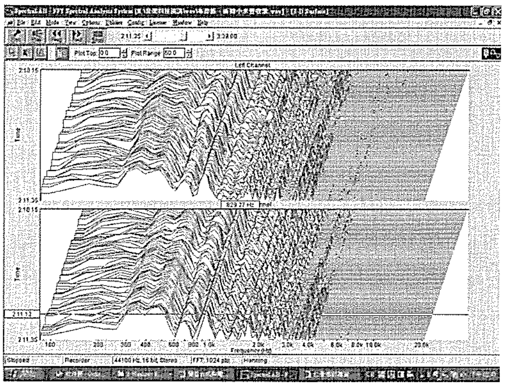


Figure 15. Using 3D Sonogram to Analyze the Octophonic Chorus "Pasibutbut" Song from the Taiwanese Aboriginal Bunun People

Figure 16 shows the timbre change of the Chinese big drum sound, by using the 3D sonogram. The low frequency spectrum distribution shows no significant or clear fundamental and higher harmonic partials, therefore the definite pitch sound can not be generated by the Chinese big drum.

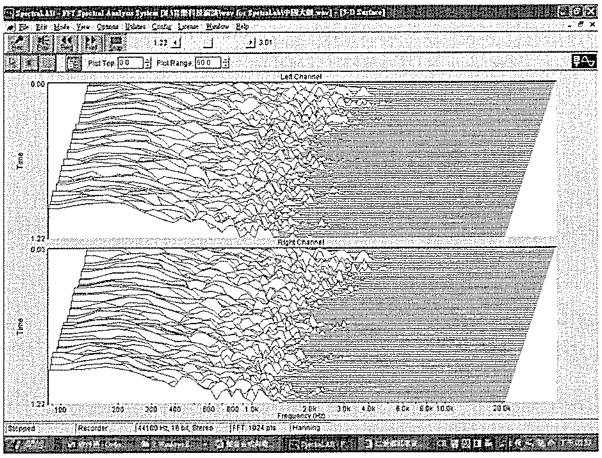


Figure 16. The Timbre Change of the Chinese Big Drum Sound by Using the 3D Sonogram

(4) The Discussion for the Multi-track Montage Composition Methodology

Using the above mentioned sound synthesis techniques, we can import the audio samples with the audio process software Audition to perform the transformations including looping, reverse, and many kind of effects, to alter the sound quality, in order to form the digital concrete music. Although the MIDI interface and the sound modules can be also used to generate the imitative sound for the acoustic instrument, we should notice that this kind of imitative sound can not provide the dynamic articulation and the vivid timbre change. Therefore the Multi-track Montage for the sound process software Audition with the audio transform techniques are the better choices for the integration with the methodology of all kind of the sound synthesis to compose the outstanding music works.

On the other hand, it is worth to learn that the compute music composition is the same as the literature writings, which needs the capability of structural thinking and process, to correlate the sound material into the text of the music works, to avoid the superfluous phrases in the writings and loosing the musical tension. As shown

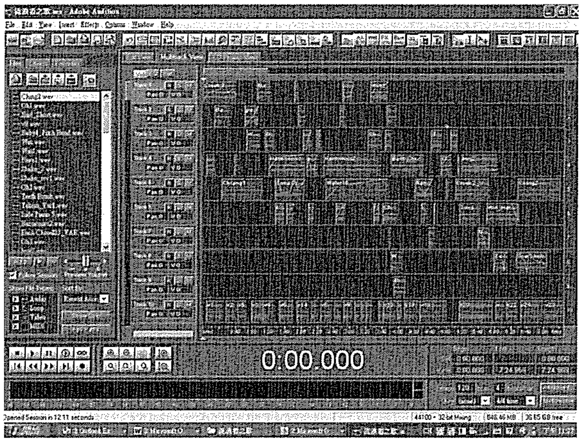


Figure 17. The Muti-track Montage for the Digital Audio Waves

in Figure 18, no mater what kind of the sound synthesis and transformation techniques are adopted, and regardless of what kind of the computer music programs are used, all of the musical parameter settings should be considered as the response of the music text. It is consistent with the organic four steps in the composition of an essay: introduction, elucidation of the theme, transition to another viewpoint and summing up, then we can allow the audio process to get in the final muti-track collage stage.

Conclusion

The fundamental acoustic and physic theories are discussed in the above text, and we can learn and practice all kind of the sound synthesis and transformation techniques by using music software programs. Only if the tight integration of music and technology with deep research, we can make a successful environment for sound synthesis, computer music composition, and digital art education. The unification of the research in the art field of computer music with the engineering development in the technology field can express the features of diversification and flexibility from the integrated technology, to apply the scientific research into the composition of digital art music. An integrated steady bridge between art and information technology can be established via the intrinsic characters of the information research and technology integration, to construct a complete

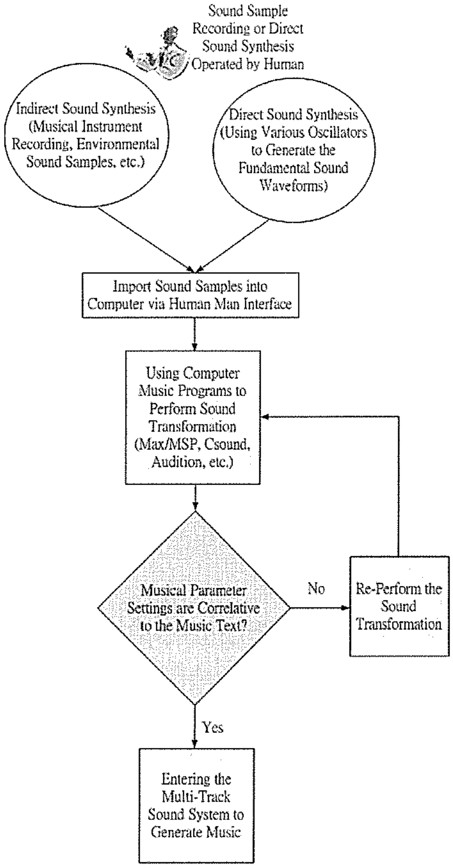


Figure 18. The Procedures of Compute Music Composition

educational environment for computer music progressively, and then turn Taiwan into a world-class place for the interdisciplinary research of art technology.

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