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SYNTHESIS OF KANTIL TONE USING THE FREQUENCY MODULATION METHOD

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Abstract

Music is a creative form of expression that utilizes sound arranged in specific patterns to create artistic works. Excessive exposure to loud sounds can harm hearing, potentially leading to hearing loss. A significant challenge in crafting musical instruments, such as the Balinese kantil, is the variability in sound produced by different artisans, resulting in inconsistent rhythms in angklung gamelan performances. This research aims to synthesize kantil sounds using digital methods to standardize the sound and ensure consistent tonal quality across instruments. The methodology involves preprocessing the audio data using Fast Fourier Transform (FFT) and Hilbert Transform to extract fundamental frequencies and signals. Frequency Modulation (FM) is then applied to synthesize the sound. The synthesized sound is evaluated using Root Mean Square Error (RMSE) to compare the original signal with the synthesized one. This process produces a kantil sound that closely mimics the original, contributing to consistency in musical performances. By offering a reliable method to standardize sound, this research supports both craftsmen and musicians in preserving the cultural integrity of Balinese music.

Keywords : Frequency Modulation, Root Mean Square Error, Fast Fourier Transform, Kantil

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INTRODUCTION

The rapid advancement of technology has significantly impacted various fields, including sound signal processing. This technology is widely used not only in sound but also in music, video, and image processing [1]. Music is a powerful form of creative expression, employing sound arranged in structured patterns to create art that pleases the listener. Among Bali's traditional musical arts, the gambelan angklung holds a significant place, often accompanying various Balinese dances. One of the key instruments in the gambelan angklung ensemble is the gangsa kantil (kantil), a metallophone with metal plates horizontally arranged on a wooden frame. Each plate produces a unique note, and when struck, it creates a distinct and harmonious sound.

Traditionally, the construction of the kantil relies heavily on the artisans' sense of hearing. However, fewer artisans are involved in making the gangsa kantil gamelan, mainly because of a lack of interest in learning the craft [2], which is partly due to the limited availability of learning resources and media.However, human hearing is limited to a frequency range of 20 to 20,000 Hz, but the kantil's sound frequencies normally vary between 5,000 and 15,000 Hz. Prolonged exposure to loud or continuous noise can harm hair cells in the ear, potentially resulting in hearing loss or deafness. The emphasis on manual auditory craftsmanship causes variances in the sound generated by different kantil instruments, resulting in rhythmic discrepancies during gambelan angklung performances. The sound produced by the gambelan needs to undergo sound synthesis using specific parameters to control it [3]. The gambelan is a musical instrument with a pentatonic scale (pelog), consisting of five notes [4]. Due to the limited number of notes, there is a possibility that the sound may become inconsistent.

To address these inconsistencies, the present research investigates the ability of digital sound synthesis to standardise kantil tones. Digital synthesis was chosen to address the variation in sound produced by different kantil craftsmen, aiming to standardize the tonal output through precise computational methods. Frequency Modulation (FM), combined with techniques like Fast Fourier Transform (FFT) and Hilbert Transform, ensures accurate reproduction of the fundamental frequency and envelope signals of the kantil. The process produces synthetic sounds that closely resemble the original, ensuring consistency in the musical performance.

Several studies investigated into the application of FM in signal processing and sound synthesis. Frequency Modulation method was used to synthesize the tone of the terompong [5], another Balinese instrument, demonstrating the effectiveness of FM in preserving the tonal integrity of traditional instruments. In this work, FM was employed to generate harmonic and non-harmonic fluctuations in sound, resulting in realistic output. The research conducted by [6] primarily investigates the use of Double Frequency Modulation (DFM) and Fast Fourier Transform (FFT) to synthesize the sound of the rindik instrument, aiming to preserve its tonal integrity in digital formats. Researchers in [7]also explores the use of the Double Frequency Modulation (DFM) method to synthesize the sound of the traditional Balinese musical instrument gerantang. The synthesis process involved calculating the fundamental frequency and signal envelope during the preprocessing stage, followed by the synthesis using DFM. In [8] FFT is used to identify the sound's frequency components, allowing them to replicate the instrument's tones accurately in the digital synthesis phase.

Several studies focused on sound synthesis such as those carried out by Sean O'Leary [9].The extracted features, such as decibel (dB) levels and frequency values, were then analyzed to ensure that the synthesized sounds closely matched the original gerantang tones. FFT method is also used in [8] to analyzed sound patterns to determine the characteristics of the sunari sound. Through FFT, key features such as amplitude and frequency are extracted, which are essential for understanding the acoustic properties of the sunari sound.

This research intends to help kantil artisans produce instruments with consistent tone quality, keeping the harmony and rhythm of gambelan angklung performances. The findings of this research have practical significance for both musicians and artisans, guaranteeing that the cultural history of Balinese music is preserved in the current era through technical developments.

METHOD

Data Collection Method

The primary data for this investigation were audio recordings of a gangsa kantil instrument played by an artisan in Mengwi, Bali. Individual tones from each kantil plate were recorded using a Samsung A71 smartphone, resulting in 56 data. 44 data were used for training, and 12 for testing. Each data contains a one-second recording of a single kantil plate stored in WAV format.

Preprocessing Method using Fast Fourier Transform (FFT) and Hilbert Transform

Preprocessing was carried out utilizing the Fast Fourier Transform (FFT) method. FFT is used in the fields of spectrum analysis and optical signal processing as well as digital filter design [10]. FFT is a technique for transforming a time-domain signal (which describes sound as it grows over time) into a frequency-domain signal (which depicts sound as a collection of different frequencies). This method was used to extract critical acoustic properties, such as the instrument's fundamental frequency and amplitude, which are required for sound synthesis [8]. Windowing was used to divide each audio stream into manageable parts. This technique applied during signal processing, particularly in Fast Fourier Transform (FFT) calculations, to reduce spectral leakage and improve the precision of frequency and amplitude estimations [11]. Following FFT, the Hilbert Transform was used to obtain the signal envelope, which helps capture the amplitude characteristics of each audio source.

In this technique, the Hilbert Transform is used to extract the envelope of the audio signals for each kantil bar. The Hilbert Transform shifts the signal phase by 90 degrees[12], allowing the researchers to extract the envelope of the signal, which includes the amplitude and phase information[13]. The envelope represents amplitude variations over time and is important for capturing the signal's dynamic properties. This information is then employed in the Frequency Modulation (FM) procedure to alter the synthesized kantil sound such that it closely resembles the original recordings. The Hilbert Transform data is modified using the fundamental frequencies acquired from the FFT technique, providing a precise depiction of the kantil's tonal properties.

The following tables, Table 1 to 4, describes the kantil audio dataset after it has undergone FFT processing to extract the fundamental frequency data.

Table 1. Kantil Traning Data (ding)

No	Tone	Audio File (wav)	Fundamental Frequency (Hz)
1	ding	ding1	646,5694
2	ding	ding2	646,8750
3	ding	ding3	646,5320
4	ding	ding5	646,8636
5	ding	ding6	646,8636
6	ding	ding7	6467289
7	ding	ding8	646,4843
8	ding	ding11	646,6836
9	ding	ding12	646,6836
10	ding	ding13	646,1397
11	ding	ding14	646,9375
Average			646,5953

Table 2. Kantil Traning Data (dung)

No	Tone	Audio File (wav)	Frequency (Hz)
1	dung	dung1	738,5204
2	dung	dung2	738,2812
3	dung	dung3	738,5204
4	dung	dung5	737,8120
5	dung	dung6	738,0319
6	dung	dung7	737,8125
7	dung	dung8	738,0516
8	dung	dung11	738,0514
9	dung	dung12	738,0515
10	dung	dung13	738,0524
11	dung	dung14	738,2812
Average			738,0427

Table 3. Kantil Traning Data Set (deng)

No	Tone	Audio File (wav)	Frequency (Hz)
1	deng	deng1	836,0969
2	deng	deng2	836,3970
3	deng	deng3	836,2500
4	deng	deng5	836,3970
5	deng	deng6	836,3970
6	deng	deng7	836,2500
7	deng	deng8	836,6370
8	deng	deng11	835,6370
9	deng	deng12	837,2500
10	deng	deng13	835,9375
11	deng	deng14	836,0969
Average			836,1057

Sound Synthesis using Frequency Modulation Algorithm

After obtaining the fundamental frequencies and signal envelopes through preprocessing, the Frequency Modulation (FM) technique was employed to generate the kantil sound. FM uses one signal (modulator) to change the frequency of another signal (carrier), resulting in rich harmonic and inharmonic partials [14].

Table 4. Kantil Traning Data Set (dong)

No	Tone	Audio File (wav)	Frequency (Hz)
1	dong	dong1	973,6328
2	dong	dong2	974,0338
3	dong	dong3	974,2647
4	dong	dong5	973,8524
5	dong	dong6	974,0625
6	dong	dong7	973,4042
7	dong	dong8	973,4042
8	dong	dong11	973,6328
9	dong	dong12	974,0625
10	dong	dong13	974,2647
11	dong	dong14	974,0625
Average			973,9182

For this method, the carrier frequency was defined by the FFT-extracted fundamental frequencies, and the modulating signal was represented by the Hilbert Transform envelope. The carrier wave has a base frequency of 1000 Hz, while the input sound has a frequency of 100 Hz. As the loudness of the input sound increases, the carrier wave's frequency also rises. When the input sound is at its loudest, the carrier wave reaches its highest frequency. When the input sound becomes quieter, the carrier wave's frequency drops to its lowest point. Essentially, the carrier wave's frequency changes in response to the volume of the input sound. Frequency Modulation is shown in equation (1)

$$f_m(t) = A\cos\ \omega_c t + \frac{\omega}{\omega}\sin\omega_m t \quad (1)$$

Where fm(t) is the frequency-modulated wave, A is the amplitude of the carrier signal, c is the angular frequency of the carrier signal, and m is the frequency of the message signal (modulator). The synthesized sound was saved in WAV format, allowing for comparison of the original and synthesized audio files.

The Implementation is shown in figure 1. The process begins with importing the recorded kantil sound files. The data is then divided into two categories: training and testing data. Each dataset is preprocessed using the Fast Fourier Transform (FFT) to extract essential features, with an emphasis on fundamental frequencies. Spectrum analysis is conducted using the Fast Fourier Transform (FFT) method [15]. Furthermore, the Hilbert Transform is employed

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to determine the best sound properties for each kantil blade, which are then used in the Frequency Modulation process.

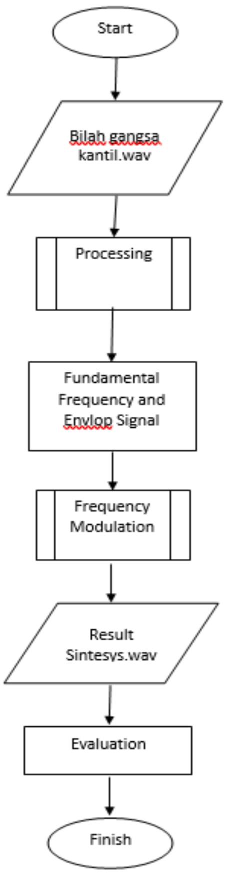

Figure 1. Synthesis Process

After preprocessing the per-bar dataset, the fundamental frequency and related signal are retrieved from each dataset. In the ensuing evaluation phase the Root Mean Square Error (RMSE) is calculated to determine the difference between the original and synthesized signal. This assessment analyzes how well the generated kantil sound matches the original. The end product is a synthesized kantil sound that closely resembles the original sound.

Evaluation using Root Mean Square Error (RMSE)

To assess the accuracy of the synthetic kantil sound, the Root Mean Square Error (RMSE) was determined. The RMSE measures the difference between the original and synthetic audio signals. The formula of RMSE as follow [16]:

$$RMSE = \sqrt{\frac{1}{n}\sum_{i=1}^{n}(y_i - \hat{y}_i)^2} \quad (2)$$

n = Number of observations
yi = Observed value
$\hat{y}_i$ = Predicted value

A lower RMSE value indicates better performance, as it reflects a smaller error or deviation [17]. Conversely, higher RMSE values suggest that the model's predictions are less accurate. Because RMSE squares the differences between predicted and observed values, it is particularly sensitive to large errors. This sensitivity makes RMSE a valuable tool for identifying models that struggle with significant deviations.

System Development and Testing

The synthetic kantil sound was integrated into a web-based system built with Python signal processing and JavaScript user interface. The system allows users to upload kantil sound files and compare them to artificial sounds. Functional testing was performed using the Black-Box testing method, which is a widely used approach in software testing to assess the overall functionality of a system without delving into its internal code structure [18]. This method focuses on evaluating how the system behaves based on the inputs provided and the outputs generated, ensuring that the software meets the required specifications and functions as expected [19]. This strategy focuses on verifying that the system performs as expected given its input and output. The primary benefit of Black Box Testing is its capacity to detect flaws, defects, or malfunctions in the system by mimicking real-world user interactions [20]. The program flow is shown in the figure 3.

The system includes two features: the tone for each bronze kantil blade and the main page display. The first stage of the system is designed to provide information on the main page, which includes a feature to play audio when clicked. According to the flowchart, when a user enters the system, they have the option to upload a WAV file. If a file is uploaded, the backend system will process it through preprocessing. Following preprocessing, the system will display the frequency and accuracy of the Kantil sound. If no kantil sound is provided by the user, the system will instead play the synthetic sound of the bronze kantil that is already integrated into the system.

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RESULT AND DISCUSSION

Data Collection Process

This research uses primary data collected through voice recordings. The recordings were made with the microphone of a realme 7i mobile phone, capturing the performance of each individual bar of the kantil. Each bar was struck alternately, and the resulting sounds were recorded using the smartphone's voice recorder. Figure 5 illustrates the process of recording these sounds. The data collection process used in this research employed the observation method, where the sound data was recorded directly while the Gangsa Kantil was played according to its base notes. Only one kantil instrument was played to obtain the dataset. A total of 56 datasets were used, consisting of 44 datasets for training data and 12 datasets for testing data. The 44 training datasets were divided into 4 datasets per Gangsa Kantil note, and the 12 testing datasets were also divided into 4 datasets per kantil note. Each dataset's audio recording file lasts for 1 second per kantil key. The recording format used is the .wav file extension

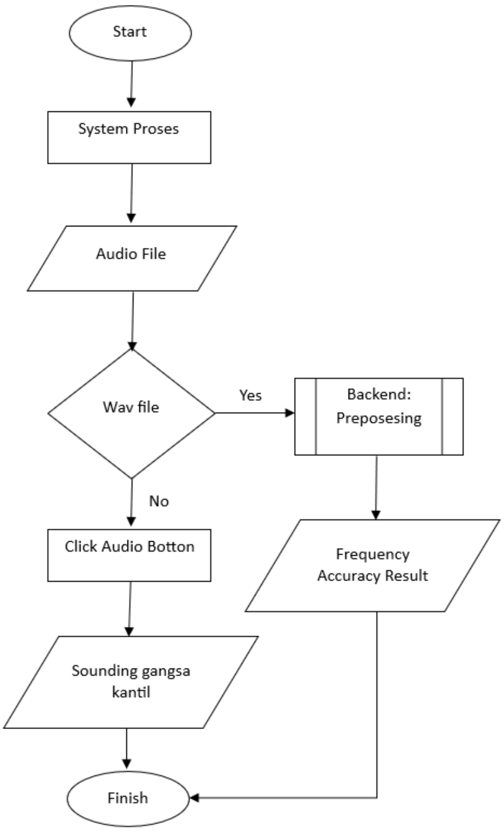

Figure 3. Application Process

Results of Preprocessing Method using Fast Fourier Transform (FFT) and Hilbert Transform

In the process of determining the fundamental frequency of each audio dataset, a data processing method using the Fast Fourier Transform (FFT) is required. The data processing involves several steps, including windowing the audio frames that have been input. The windowing technique simplifies data processing by dividing the audio frames, making it easier to identify the peak points of frequencies within the frame range.. After the audio undergoes the windowing process, it is then processed computationally to determine the frequency peak points more accurately. Below is the Gangsa Kantil audio dataset that has undergone the Fast Fourier Transform process for determining its fundamental frequency data.

Figure 5. Kantil Recording Process

Table 5. Kantil Data Testing

No.	Tone	File nama (wav)	Frekuensi (Hz)
1.	ding	ding4	646,1397
2.	ding	ding9	646,4843
3.	ding	ding10	646,6836
	Average		646,4359
4.	dung	dung4	737,8125
5.	dung	dung9	738,5204
6.	dung	dung10	737,8125
	Average		738,0484
7.	deng	deng4	835,6334
8.	deng	deng9	836,2500
9.	deng	deng10	836,2500
	Average		836,04449
10.	dong	dong4	974,2647
11.	dong	dong9	974,0625
12.	dong	dong10	973,8520
	Average		974,0597

The Hilbert transform process is used to determine the input Kantil audio dataset [5]. This data consists of an array of amplitudes for each audio dataset. The data processed through the Hilbert transform will be modulated using the frequencies obtained from the Fast Fourier Transform (FFT) process. After applying the Hilbert transform for each audio dataset is derived. The following provides an overview of the results for each audio dataset

1. Ding envelove ilustration 1.1.wav
Here is the envelope representation of the gangsa ding1.wav audio dataset obtained through the computational process of Hilbert transformation, as shown in the figure 7.

2. Dung envelove illustration 2.1
Here is the envelope representation of the gangsa dung1.wav audio dataset obtained through the computational process of Hilbert transformation, as shown in the figure 8

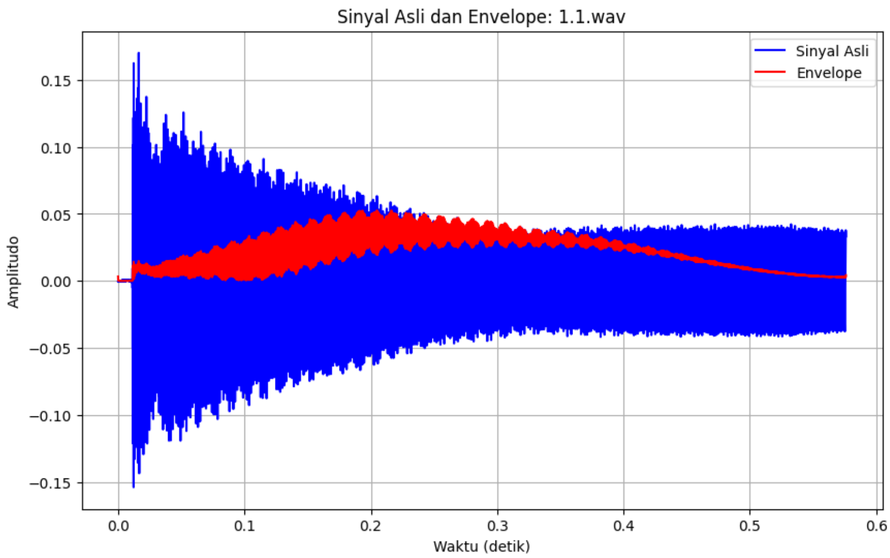

Figure 7. Ding Envelope Illustration

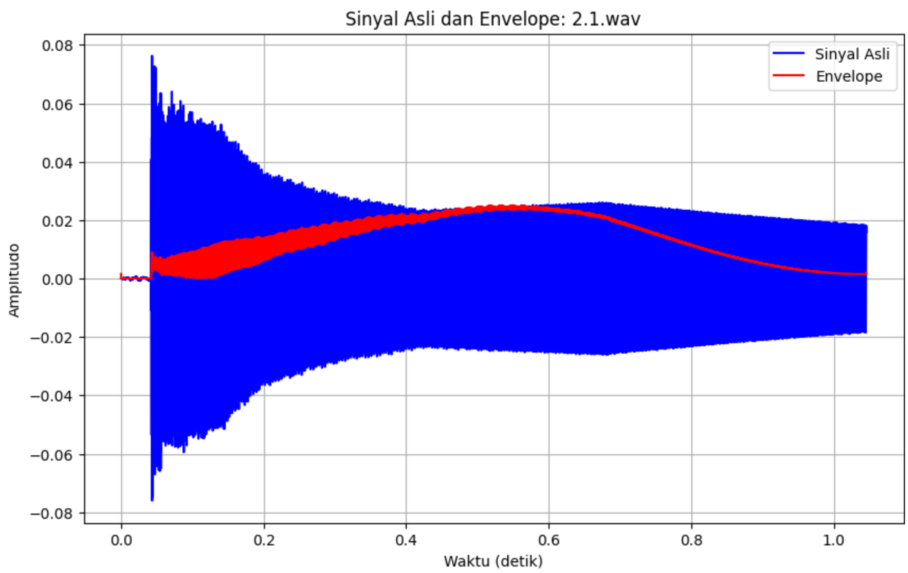

Figure 8. Dung Envelope Illustration

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3. Deng Envelove Illustration 3.2
Here is the envelope representation of the gangsa deng2.wav audio dataset obtained through the computational process of Hilbert transformation, as shown in the Figure 9.

4.Dong envelove illustration 4.1
Here is the envelope representation of the gangsa dong1.wav audio dataset obtained through the computational process of Hilbert transformation, as shown in the Figure 10.

Results of the Sound Synthesis Process using Frequency Modulation Algorithm

Testing of the results of the synthesis process was obtained by searching for basic tone frequencies from both datasets, namely the synthesized audio dataset and the audio data testing dataset. The basic frequency of synthesized audio tones and the audio training data dataset. The level of success of the results of the synthesis process can be seen from the comparison between the average frequency of the synthetic data and the average frequency of the tones in the testing data. the difference is close to zero, and the notes ding, dung, deng, dong in synthesis using a frequency modulation algorithm are close to the original sound.

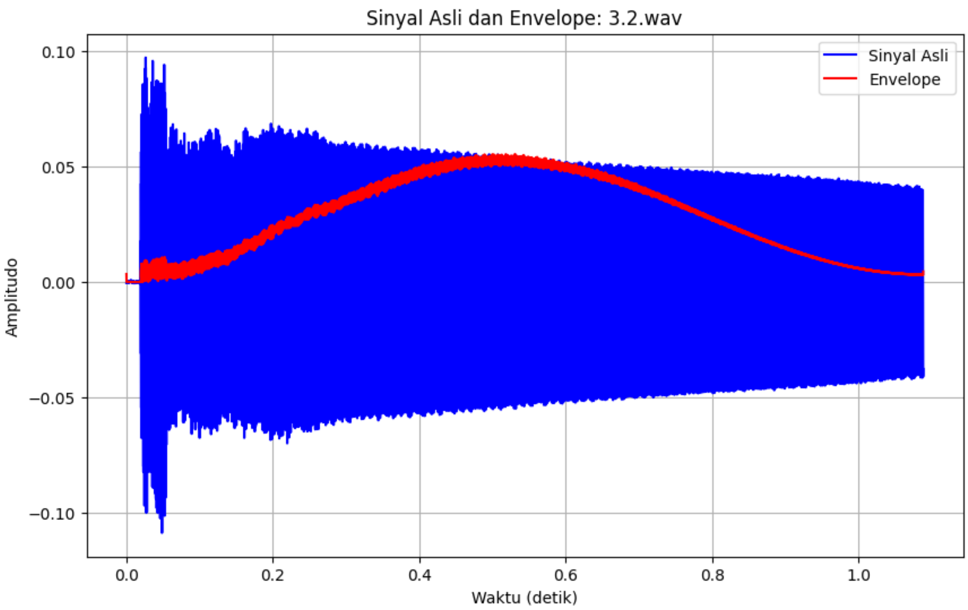

Figure 9. Deng Envelove Illutration

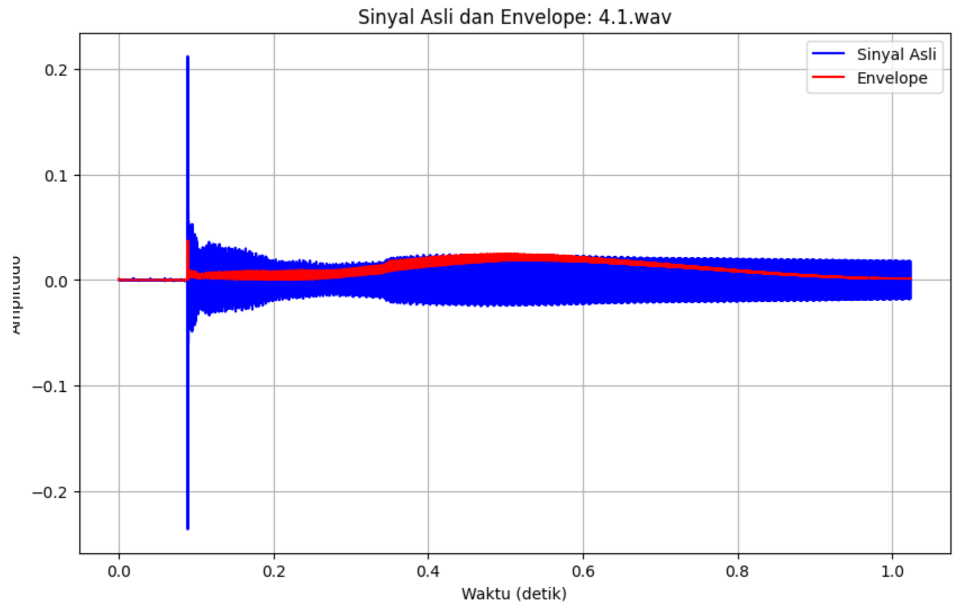

Figure 10. Dong Envelope Illustration

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The accuracy level is determined by the difference between the synthesized frequency and the original frequency. The smaller the difference, the better the results of the synthesis process. In other words, accuracy improves as the difference between the two frequencies decreases. Conversely, accuracy worsens if the difference increases.is used in calculating the difference value as a calculation of the results of testing the sound synthesis process and uses RMSE to determine the accuracy of sound synthesis.The results can be seen in Table 6.

Results of Evaluation using RMSE

The evaluation of the synthesis process using Root Mean Square Error (RMSE) involves comparing the basic tone frequencies from two datasets: the synthesized audio dataset and the recorded audio dataset. By comparing the basic tone frequencies of the synthesized audio with those of the recorded data, the success of the synthesis process can be assessed. This is done by calculating the average frequency of the tones in both datasets. A comparison of these average frequencies will yield a value close to zero if the synthesized tones closely match the original tones. This approach helps determine how closely the synthesized frequencies align with the original frequencies.

Results of System Developmen and Testing

Figure 6 shown the application interface. Web design was developed using Python and JavaScript programming. The system utilizes audio data and offers two features: tone per bronze kantil blade. In the initial stage, the system is designed to display information on the main page, featuring an option to output audio when clicked. Additionally, users can input a sound file to obtain the fundamental frequency value and assess the accuracy between the input data and the sound synthesis data.

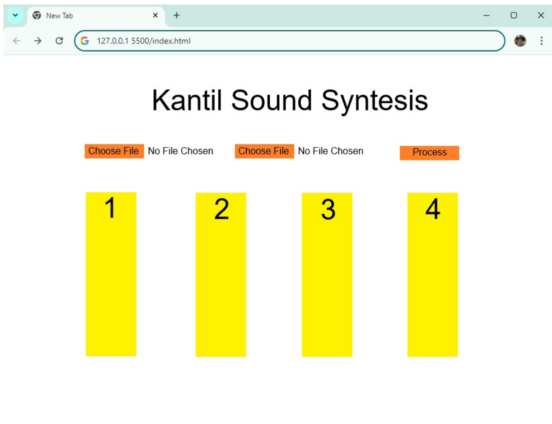

Figure 6. Kantil Sound Systesis Apllication

The expected outcomes and system testing results are consistent, as shown in Table 8, confirming that the feature is functioning properly. Integration Testing evaluates the interactions between various components within the application to ensure accurate sound synthesis. This involves testing sound data with different variations to assess overall component performance. The results from Integration Testing are demonstrated through a range of scenarios tested on the web-based system.

Table 6. Sound Synthesis Testing Results

Tone	Synthesis Frequency (Hz)	Testing Data Frequency (Hz)	Difference	RMSE	Accuracy
1	646,6836	646,6365	0,6907	0,6907	99,85%
2	738,5204	737,8125	0,7079	0,7079	99,90%
3	835,6334	836,6120	0,9785	0,9785	99,88%
4	974,8086	973,8520	0,9566	0,9566	99,90%

Table 7. System Functionality Black Box Testing Results

No	Scenario	Plan	Expected results	Result Testing
1.	Box blade 1	Click blade Tone 1 (ding)	Makes the sound of a kantil blade with a tone ding	Normal
2.	Box blade 2	Click blade Tone 2 (dung)	Makes the sound of a kantil blade with a tone dung	Normal
3.	Box blade 3	Click blade Tone 3 (deng)	Makes the sound of a kantil blade with a tone deng	Normal
4	Box blade 4	Click blade Tone 4 (dong)	Makes the sound of a kantil blade with a tone dong	Normal

Table 8. Integration Testing Results

No	Scenario	Plan	Expected results	Result Testing	Conclusion
1.	Box blade 1	Click blade Tone 1 (ding)	Makes a sound similar to a kantil blade on the testing data with a ding tone on the web	As expected	Valid
2.	Box blade 2	Click blade Tone 2 (dung)	Makes a sound similar to a kantil blade on the testing data with a dung tone on the web	As expected	Valid
3.	Box blade 3	Click blade Tone 3 (deng)	Makes a sound similar to a kantil blade on the testing data with a deng tone on the web	As expected	Valid
4	Box blade 4	Click blade Tone 4 (dong)	Makes a sound similar to a kantil blade on the testing data with a dong tone on the web	As expected	Valid

The results align with the desired expectations, with system testing showing a total accuracy of 99% for each sound per blade. The sound synthesis closely matches the original kantil sound, and the testing process involved kantil artisans. Therefore, the system is deemed suitable for users and has performed according to its intended functionality.

CONCLUSION

Based on research that has been carried out on the application of the frequency modulation algorithm to create a synthetic sound of kantil as a digitalization of culture, in determining the results of the synthetic sound of kantil, which will be used for accuracy using a modulation index value of 30, which is a value that is almost similar to the original sound of gangsa. kantil and determining the modulation index involves the kantil craftsman to determine the appropriate modulation index value. The sound produced from synthesis using the FM algorithm shows similarities to the original sound of kantil (voice 1 accuracy: 99.85%, voice 2 accuracy: 99.90%, voice 3 accuracy: 99.88%, and voice 4 accuracy: 99.90%). This indicates that the chosen technical approach is effective in reproducing the sound characteristics. And for each frequency obtained from the synthesis method used for Sound 1: 646.6837 Hz, Sound 2: 738.5204 Hz, Sound 3: 835.6335 Hz, and Sound 4: 974.8087 Hz. With the results of the tests that have been carried out, it can be concluded that the system that has been created can be used by users and craftsmen to assist in making kantil and is an easy way to introduce the kantil musical instrument to the public.

While the present research was effective in synthesizing gangsa kantil sounds with FM, there are various areas for further research. One significant area for improvement is the dataset, which should be expanded to include additional types of gangsa kantil instruments. Furthermore, future work could concentrate on implementing more complex sound generation techniques or upgrading the web application's functionality to enable real-time sound manipulation. Further research could look into adapting this synthesis process to other traditional instruments to aid in larger cultural preservation efforts.

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