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HIGH PERFORMANCE GAMELAN ANALYZER USING ADAPTIVE WAVEFORM PATTERN MATCHING

[1]YOYON K. SUPRAPTO, [2]VINCENTIUS E. PRADHANA
[1,2]Electrical Engineering Department, Institut Teknologi Sepuluh Nopember, Surabaya, Indonesia.
E-mail: [1]yoyonsuprapto@ee.its.ac.id , [2]vinc3nt@elect-eng.its.ac.id

ABSTRACT

Exploration of the eastern music such as gamelan is very rare, so its development is far lagged to western music. Estimation of fundamental frequencies and estimation of the envelope are needed to determine the performance of traditional music instrument. We applied Waveform Adaptive Pattern Matching (WAPM) to estimate fundamental frequency which is detected with limited frequency band.The fundamental frequency is estimated by very short recorded gamelan music signal of 45 ms, with accuracy of 0.25 Hz. Besides the fundamental frequency estimation, the invention is useful gamelan tuning the signal envelope.
This method is implemented in Android applications to analyze a signal gamelan, thus enhance the portability of the application.
Keywords: *Gamelan Music, Fundamental Frequency, Waveform Adaptive Pattern Matching, Minimum Absolute Error, Android.*

1. INTRODUCTION

Gamelan music instrument as a traditional music of Indonesia is a music instrument that is need to be preserved in its existence as one of the national's heritage. Fundamental frequency estimation of a sound signal, can be done by using tuner, or well-known as an electronic tuner. Gamelan music instrument are using pelog (hexatonic) and slendro (pentatonic) scales [1]. There is no electronic tuner that is based on those two music scale. Some researchers are trying to find the similarities of pelog-slendro scales and diatonic scales as listed in Table 1 and Table 2 [2].

TABLE 1 Similarity Of Pelog And Diatonic Scales

Pelog Scale	Frequency (Hz)	Diatonic Scale
1	599	D#-23
2	638	E-20
3	698	F+10
4	798	G+16
5	830	G#+35
6	924	A'+44
7	1008	B'+24

The development of gamelan music is very rarely done by researchers and musician. This lack of development has left gamelan music behind with western music. By doing development in the field of gamelan music will hopefully catch up gamelan music with western music. Furthermore, the development is done to restore the greatness of gamelan music as in his era (17-18th century) [3].

TABLE 2 Similarity of Slendro and Diatonic Scales

Slendro Scale	Frequency (Hz)	Diatonic Scale
1	530	C+4
2	608	D#-7
3	698	F-20
5	804	G+11
6	924	A#'-4
1'	1056	C'+5

Sindusawarno, an expertise in gamelan, had also conducting research about similarity of Pelog-Slendro scales and Diatonic scale. Furthermore, he was also conducting research about cents division of gamelan [1], as shown in Fig. 1. This cents division can also be applied as standardization reference in manufacturing gamelan.
The construction of a gamelan equipment can be seen in Fig. 2. When the blades hit it will produce a sound source, the absorber is applied as silencers and resonator acts as echo maker.
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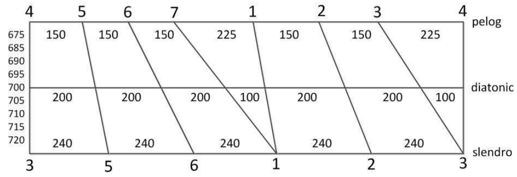

Fig 1.Cent Division For Pelog, Slendro And Diatonic.

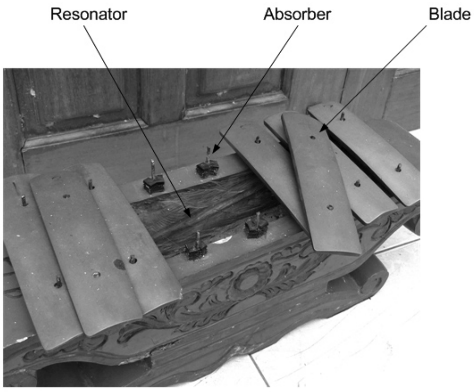

Fig 2. Construction Of A Gamelan Equipment.

The characteristic of gamelan is that one instrument in a gamelan instrument represents one octave. Octave that is located below or above the instrument is possessed by the other. Example: Saron instrument has its own specific frequency characteristics. One octave above Saron, contained in the Peking instrument, while an octave below the Saron contained in the Demung instrument.

Each gamelan is made with various manufacturing method, depending on the manufacturer. Gamelan is manufactured manually, which is worked by a craftsman. A gamelan, from one craftsman to another craftman can be very different. It is because there is no standardization in manufacturing gamelan instrument. Craftsmen, in manufacturing gamelan, is greatly rely on the hearing sensitivity of gamelan's craftsmen themselves, while the hearing sensitivity of each gamelan craftsmen may different from one to another.
The minimum, maximum and average fundamental frequency can be seen on Table 3 and Table 4, they can be seen that from 25 on the different set of gamelan, suggesting that the characteristics of gamelan on the market can be very different from one gamelan to another.
Hence, we introduce a Waveform Adaptive Pattern Matching method. This method is to perform cross production on the observe signal and the reference signal. Observe signal in the form of a sound signal, and then compared the similarity with the reference signal. Reference signal is a signal generated automatically with parameters, such as frequency, amplitude, phase and the length signal. Those reference signal parameters are varied so that the shape is similar to the input signal.

Table 3 Maximum, Minimum And Average Value Of The Pelog Fundamental Frequency On Several Gamelan Instruments Demung, Saron And Peking

Instrument Name		Fundamental Frequency (Hz)					
		Pelog Notation					
		1	2	3	4	5	6
Demung	AVG	299	321	349	410	440	470
	MIN	294	316	344	404	435	464
	MAX	300	323	352	412	442	472
Saron	AVG	597	646	697	822	887	944
	MIN	590	637	688	815	867	935
	MAX	608	655	706	827	908	953
Peking	AVG	1205	1301	1411	1645	1774	1899
	MIN	1189	1276	1390	1595	1739	1868
	MAX	1224	1320	1431	1658	1839	1951

Table 4 Maximum, Minimum And Average Value Of The Slendro Fundamental Frequency On Several Gamelan Instruments Demung, Saron And Peking

Instrument Name		Fundamental Frequency (Hz)				
		Slendro Notation				
		1	2	3	5	6
Demung	AVG	268	307	354	406	465
	MIN	264	303	352	403	463
	MAX	271	311	356	410	467
Saron	AVG	534	610	704	805	925
	MIN	515	589	681	775	897
	MAX	544	621	716	820	943
Peking	AVG	1065	1219	1397	1618	1863
	MIN	1038	1195	1363	1594	1851
	MAX	1115	1249	1434	1665	1884

By using only a few numbers of samples, we can estimate the fundamental frequency. By using

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Saron Slendro, for resolution of 1 Hz, samples of 45 ms (2000 samples) are required. For resolution of 0.25 Hz, samples of 204 ms (9000 samples) are required.

The proposed Waveform Adaptive Pattern Matching is implemented in software that can be operated in Android. The program can be used to analyse the gamelan sound signals, for the purpose of tuning for the manufacture of gamelan music instrument itself. By implementing on Android, we could invent a portable electric tuner for gamelan. Besides the fundamental frequency estimation, the invention is useful gamelan tuning the signal envelope. It is very usefull for gamelan craftments to create better gamelan.

The paper is organized as follows. Part II discusses the related works Fourier Transform. Section III discusses the proposed method, including the algorithm of the proposed method. Section IV discusses an experiment conducted to determine the performance of the proposed method, including the comparison between the estimated fundamental frequency Fourier Transform method and the proposed method. Section V concludes the paper.

2. RELATED WORKS

2.1 Fourier Transform

In simple terms, Fourier Transform changing waves (sound, images, etc.) from the time domain to the frequency domain. The results of this method, forms the Fourier Transform frequency spectrum. The results of the Fourier Transform can be restored to the original signals by the reversal process (inverse) [4-7].

DFT, also known as the Discrete Fourier [7], the Fourier Transform is performed on a discrete signal. DFT signal processing series of discrete values, then each discrete signal is calculated using the Fourier Transform to produce real and imaginary components. DFT itself can be formulated as shown in Eq. (1) and (2)

DFT Equation :

$$X(k) = \sum_{b=0}^{B-1} x(n) W_B^{kb} \quad (1)$$

The W_N is formulated as Eq. (2) :

$$W_B = e^{-j2\pi/B} = \cos\frac{2\pi}{B} - j\sin\frac{2\pi}{B} \quad (2)$$

Where B is length of the window, : value of sequence position

In order to speed up the process of calculating the DFT, the algorithm in FFT (Fast Fourier Transform) is introduced. One commonly used FFT algorithm, introduced by Cooley and Tukey. Cooley-Tukey algorithm for the FFT is done by decomposing Fourier Transform or breaks the process down into smaller, then combine them into a total transformation [6].

The method used in popular electronic tuner is the FFT[4]. Another method that can be used is the FFT by combining with BPF created with Hamming Window, to eliminate noise [5]. However FFT has disadvantages in terms of frequency resolution and the number of samples of the signal.

Frequency resolution, Rf, is the resolution of fundamental frequency estimation. Suppose that the system has a frequency resolution of 2 Hz, the system can discriminate the signal between 4 and 6 Hz, but can not discriminate signal between 4 and 5 Hz [3]. Rf can be calculated by the Eq. (3).

$$Rf = \frac{fs}{n} \quad (3)$$

where fs is the sampling frequency and n is the sample length.

It may be said, to get a high resolution of frequency estimation by using the FFT [6,7], the high number of samples are needed as well. This is the weakness of FFT methods. To overcome the weakness of the FFT method that requires considerable amount of sample to obtain a high resolution.

2.2 Short-Time Fourier Transform (Stft)

STFT is an enhancement of Fourier Transform to detect signals that are not stationary (change of frequenct at any given time). In STFT, the signal is divided into parts (segments) is very small, where the segment is assumed to be stationary signals. The width of this segment of the specified window function used [8]. Window can be an Hanning or Gaussian [9] window. This window is then shifted from the starting point to the end point signal signal. Then the Fourier Transform is applied to the signal inside the window.

STFT is formulated as shown in Eq. (4)

$$y(t) = \int_{-\infty}^{\infty} x(t) w(\tau - t) e^{-j2\pi ft}\, dt \quad (4)$$

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Where $y(t)$ is result of STFT, $x(t)$ is signal that is going to be processed, $w(\tau - t)$ is the shifted STFT window by controlling τ, $e^{-j2\pi ft}$ is the Fourier Transform process to convert to the frequency domain.

The window of time *w (t)*, which is a limitation of the STFT fixed that caused the results of time-frequency resolution remains as well. Therefore, if the window is used narrowly, the time resolution will be better, but the frequency resolution would be bad. If the window that is used is narrowed down, then the frequency resolution will be better, but the time resolution would be bad. Therefore, the results of the STFT will depend of the size of the window is used.

3. PROPOSED METHOD.

The block diagram of this proposed method can be shown in Fig. 3. Gamelan sound is recorded as analog signal. Therefore, in order to be further processed, it needs to be converted into a digital signal. The analog signal is sampled with a sampling frequency of 44100 Hz. The process of sampling by the Analog to Digital Converter (ADC) is controlled by a processor.
Signal is obtained, then made to estimate of the fundamental frequency and its envelope. It is necessary to compare the reference signal with the signal at the fundamental frequency estimation and shape the envelope.

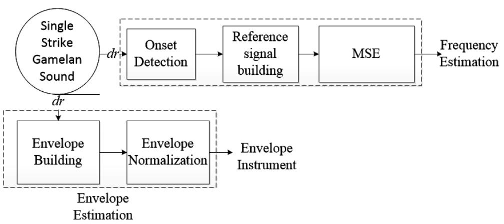

Fig. 3 General schematic of Research Methodology

Reference signal as comparator against observed signal is required, either in frequency estimation or in envelope estimation.

3.1 Reference Signal Building

This sub-chapter will explain about construction of comparison envelope, which will be used in the developed application. This envelope is used to analyze the damping characteristic of Balungan instruments (Demung, Saron, Peking)

Signal envelope depicting the outline of a signal. In other words, envelope is an imaginary line of signal waveform [11]. The envelope is constructed by absoluting signal, and then we create the border line of the signal. Envelope can also be made by sampling a few part of signal, for example by sampling the maximum value of the signal by using Eq. (5) from each N absoluted signal sample, which will produce envelope waveform on Eq. (6).

$$x_{n}(i) = \frac{x(i)}{\max(x(i))} \quad (5)$$

where x_n is normalized signal x.

$$env(i) = max_{j=((i-1)*N+1}^{i*N} x_n(j) \quad (6)$$

where *env* is envelope of the signal

This comparison envelope is constructed by observing the damping characteristic of 25 sets of gamelan. From this observation, we will get the damping model of each instrument.

The construction of comparison envelope is carried out by averaging each normalized envelope sample, n shows the instrument number of Demung, Saron and Peking and *q* shows set number of gamelan. Comparison envelope generation $\left(\overline{env}(n,i)\right)$ for each n instrument can be seen on Eq. (7).

$$\overline{env}(n,i) = \frac{\sum_{q=1}^{G} env(n,q,i)}{G} \quad (7)$$

where G is number of set of gamelan for each gamelan instrument, n, Demung, Saron or Peking.

After we obtain the envelope series of each instrument, then exponential regression is applied to obtain the damping model of $\overline{env}(n,i)$ for each n instrument of Demung, Saron and Peking. This exponential regression will produce the α and β that can be determined with Eq. (8) and (9)

$$\overline{ENV}(n,i) = \alpha(n)e^{\beta(n)i}$$

$$\log\left(\overline{ENV}(n,i)\right) = \log\left(\alpha(n)e^{\beta(n)i}\right) = \log\left(\alpha(n)\right) + \beta(n)i$$

$$\overline{ENV'}(n,i) = \log\left(\overline{ENV}(n,i)\right)$$

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$$\alpha'(n) = \log(\alpha(n))$$

$$ENV'(n,i) = \alpha'(n) + \beta(n)i$$

$$\beta(n) = \frac{K\sum_{k=1}^{K} k \times \bar{e}(n,k)}{K\sum_{k=1}^{K} k^2 - (\sum_{k=1}^{K} k)^2} - \frac{\sum_{k=1}^{K} k \sum_{k=1}^{K} \bar{e}(k,n)}{K\sum_{k=1}^{K} k^2 - (\sum_{k=1}^{K} k)^2} \quad (8)$$

$$\alpha'(n) = \frac{\sum_{k=1}^{K} \bar{e}(k,n) - \beta(n)\sum_{k=1}^{K} k}{K} \quad (9)$$

where K is the sample length of $\overline{ENV}(n)$ from exponensial regression. Then, we can obtain the damping model (comparison envelope), which is shown on Eq. (10) , (11) and (12).

i. Demung :

$$\overline{ENV}(1) = 0.8223e^{-1.194\times10^{-5}x} \quad (10)$$

ii. Saron :

iii. $$\overline{ENV}(2) = 0.7693e^{-1.03\times10^{-5}x} \quad (11)$$

iv. Peking :

v. $$\overline{ENV}(3) = 0.7663e^{-1.709\times10^{-5}x} \quad (12)$$

From previous damping model, the reference signal that is used on AWPM, reference signal can be shown in Eq. (13), (14) and (15).

i. Demung:

$$sr_d = A * \sin\left(2 * phi * \left(\frac{f}{fs}\right) * x + \theta\right) * 0.8223e^{-1.194\times10^{-5}x} \quad (13)$$

ii. Saron:

$$sr_s = A * \sin\left(2 * phi * \left(\frac{f}{fs}\right) * x + \theta\right) * 0.7693e^{-1.03\times10^{-5}x} \quad (14)$$

ii. Peking:

$$sr_p = A * \sin\left(2 * phi * \left(\frac{f}{fs}\right) * x + \theta\right) * 0.7663e^{-1.709\times10^{-5}x} \quad (15)$$

where sr_d, sr_s, sr_p are reference signals of Demung, Saron and Peking, θ is phase , f is fundamental frequency of reference signal

3.3 Onset Detection

The initial step is detecting onset. The function of detecting onset is to determine the beginning of the sound signal. Furthermore, by obtaining the onset position, we can omit the unnecessaretection is done by detecting each value of *dr* record data, starts from the beginning, that is *dr(i)* in which *i=0* until the sample length, where i is the sampled signal number. From the iteration process, if the value of *dr(i)*, where *dr(i)* more than threshold *thr,* this onset value is used and applied to cut the recording data.

Onset detection performed to remove portion of the signal which does not contain the sound of the gamelan. In simple terms, onset detection is done by looking at the sequence values $X_i(t)$ from the initial position to the end. If magnitude of $X_i(t) \geq$ 0.03 is found, then the value of t is considered as the onset (see Fig. 2). We are assuming that magnitude of $X_i(t)$ below 0.03 does not contain the real signal. Signal sequence $X_i(t)$ before t will not be used, because they do not contain the sound of gamelan.

3.3.1 Frequency Estimation, using Adaptive Waveform Pattern Matching

The next process is to create the reference signal to estimate the frequency. The diagram of reference signal generation can be seen on Fig. 5. Reference signal (*sr*) as the comparator between input signal (*dr*) is the recorded data. This data is then compared with reference signal whose parameter is varied. The parameters that are used : frequency, phase and amplitude. For each parameter changing, the Minimum Absolute Error (MAE) will be calculated, against input signal.

A combination of reference signal that give the smallest MAE value, will be considered as parameter combination that is similar to input signal parameters. Therefore, we can said that the frequency parameter of signal input is similar to frequency parameter of signal reference. The diagram of MAE calculation can be seen on Fig. 5.

3.3.2 Mean Absolute Error

As the name implied, MAE is the average value of absolute error of two different datas.

On AWPM, MAE is used to measure the resemblance of observed signal *dr(n)* against reference signal *sr(n)*. MAE on AWPM is shown on Eq. (16), as follows :

$$MAE = \frac{1}{l}\sum_{n=0}^{l} |dr(n) - sr(n)| \quad (16)$$

where l is the sample length of *dr*(*n*) and *sr(n)*.

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Signal length of *dr(n)* and *sr(n)* have to be the same. If one of the signal, either *dr(n)* or *sr(n)*, have different length, then one signal have to be cut until we obtain the same length for each signal.

MAE calculation is started by determining the length (*ps*) of recorded signal *dr*. Then we will calculate the difference (*h*) between *dr* and *sr*. the MAE value is obtained from the division of *h* and *ps*. After we obtain the MAE value, the system then saves the parameters that create the value of MAE. On the next iteration, if smaller MAE is produced, the existing MAE value and saved parameter is replaced with the new one. Otherwise, there are no replacement of the saved data.

4. EXPERIMENT

The experiments were conducted in two phases. The first phase, is estimating the frequency using the FFT frequency parameters. The second phase, is estimating the frequency parameter using the proposed method. Input signal is in the form of WAVE files (*. wav). Each file WAVE file has a sampling frequency of 44100 Hz. The sound of gamelan instruments used in this test are Saron Slendro, Demung Slendro and Peking Pelog. To abbreviate, using name of SaronSl , DemungSl and PekingPl.

The fundamental frequency of each WAVE file will be first estimated using FFT. FFT window length of 44100 is used to match the sampling frequency which is used in WAVE file. This is done to get the value of the frequency with an accuracy of 1 Hz, 0.5 Hz and 0.25 Hz. The results of these estimated frequencies using FFT method, is the benchmark (target) for testing frequency estimation using the proposed method.

Then we could perform the frequency estimation by using the proposed method. The experiment was conducted to determine the optimal value of the length of sample that can be used to estimate the parameters of the sound signal with a specific frequency. First, we developed an Android application (*.apk) using Android SDK (API 15), by implementing the proposed method. Then we use the application to estimate the fundamental frequency.

Gamelan sound file which we have obtained before is played through the loudspeaker. The sound that comes out from loudspeakers then recorded and analyzed to obtain the parameters. The position where the graph is relatively stable is called first minimum sample.

4.1 ADAPTIVE WAVEFORM PATTERN MATCHING

The next process is to create the reference signal to estimate the frequency. The diagram of reference signal generation can be seen on Fig. 5. Reference signal (*sr*) as the comparator between input signal (*dr*) is the recorded data. This data is then compared with reference signal whose parameter is varied. The parameters that are used : frequency, phase and amplitude. For each parameter changing, the Minimum Absolute Error (MAE) will be calculated, against input signal.

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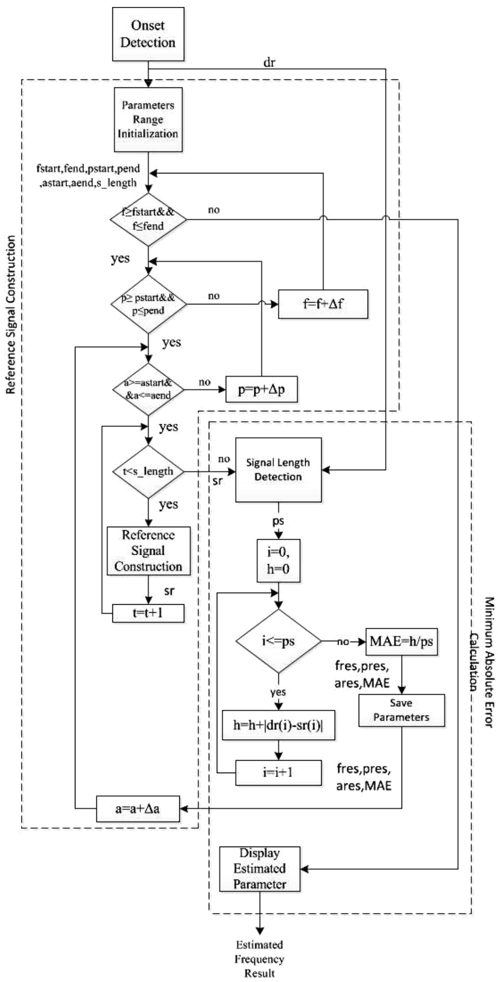

Fig. 5. Schematic Detail of frequency estimation

The fundamental frequency of each WAVE file will be first estimated using FFT. FFT window length of 44100 is used to match the sampling frequency which is used in WAVE file. This is done to get the value of the frequency with an accuracy of 1 Hz, 0.5 Hz and 0.25 Hz. The results of these estimated frequencies using FFT method, is the benchmark (target) for testing frequency estimation using the proposed method.

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Gamelan sound file which we have obtained before is played through the loudspeaker. The sound that comes out from loudspeakers then recorded and analyzed to obtain the parameters. The position where the graph is relatively stable is called first minimum sample.

5.1 Adaptive Waveform Pattern Matching

AWPM method is done by comparing the recorded signal and the reference signal. Signal is formed by varying the parameters, such as : frequency, phase and amplitude. Frequency changes is done according to the type of instrument listed in Table 3 and 4. Then we determine the minimum frequency at which the MAE occurred, as shown in Fig. 6.

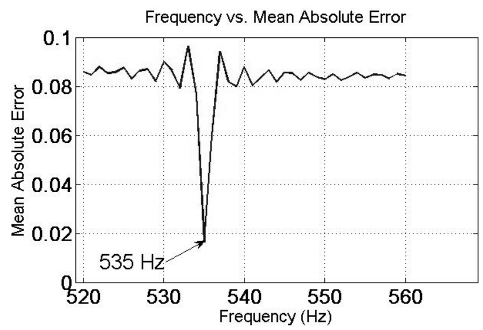

Fig. 6. Changes of MAE against fundamental frequency of reference signal.

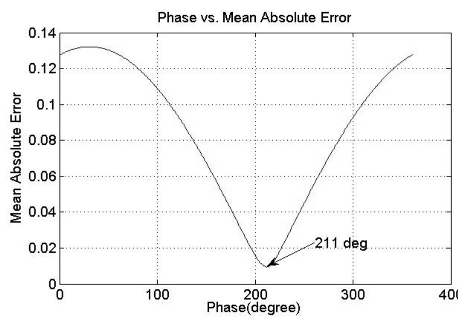

Fig. 7 Adaptive Changes in the phase of the reference signal to find the minimum MAE

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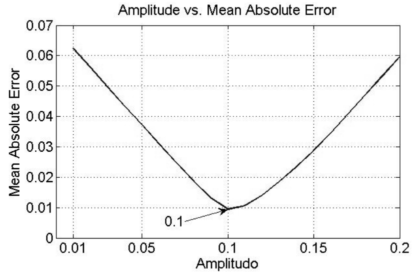

Fig. 8 Adaptive Changes in amplitude of signal reference to find a minimum of MAE.

TABLE 5. MINIMUM SAMPLE LENGTH OF REFERENCE SIGNAL FOR EACH INSTRUMENT

	Minimum sample length of signal reference (samples)		
Instrument name	Accuracy 1Hz	Accuracy 0.5Hz	Accuracy 0.25 Hz
DemungSlendro1	12000	28667	48667
DemungSlendro2	9333	37333	60667
DemungSlendro3	13000	34333	30333
DemungSlendro5	8333	22333	52333
DemungSlendro6	8667	18333	67000
DemungPelog1	6000	24600	34600
DemungPelog2	9000	15800	21400
DemungPelog3	11400	13800	25600
DemungPelog4	11200	23600	31800
DemungPelog5	8600	18400	39200
DemungPelog6	15200	26600	33000
DemungPelog7	9200	20600	38600
SaronSlendro1	9600	28933	61267
SaronSlendro2	10400	31733	52800
SaronSlendro3	11400	21933	27067
SaronSlendro5	12533	17467	25800
SaronSlendro6	17333	21267	34333

Sample length of reference signal in estimating frequency using AWPM is the minimum sample length of reference signal, so that the system can estimate range frequency parameter correctly. Minimum Sample length of each instrument can be seen on Table 5 for each resolution. If the sample length is too short, the system will unable to estimate the frequency parameter. If the sample length is too long, the system will take some time to estimate the frequency.

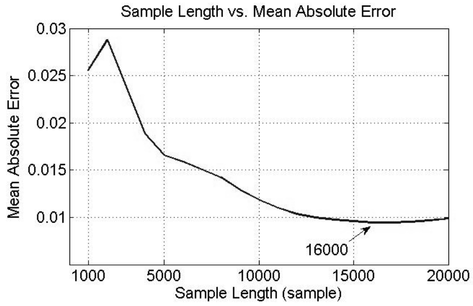

Fig. 9 adaptive Changes in length of reference signal to find to a minimum MAE

We are comparing 3 methods of estimating frequency, using FFT, AWPM and using gString. gString is an application to tune the musical instrument. Comparison of those methods, in accuracy of 1 Hz can be seen on Table 6.

From Table 5, it can be seen that the stable condition of estimating frequency, ranged from 2000 up to 7000. Not only stable, but also the system can estimate the frequency correctly. This condition, where the system is stable and able to estimate the frequency correctly, is called minimum sample length. This minimum sample length is different according to the accuracy and estimated frequency

To report an optimal frequency to be achieved, the system will display the results of frequency estimation through AWPM method, along with the average value of the frequency range for each instrument (Table 3 and Table 4). If the frequency value is lower / higher than the frequency range should be, then it would be recommended to thicken or thin the blade (if made of bronze), and bend or straighten the blades (if made of iron).

5.2 Envelope Analysis

To analyze the echo of the sound of gamelan instruments that are used, the envelope analysis is carried out. Envelope is made by getting maximum value out of every N sample of the absolute input signal values. Envelope result of recording process will be compared with the envelope comparison for each musical instrument. Th5.2ere will be two plots envelope images in one screen.

Algorithm:

1. Absolution the input signal value
2. Obtain the maximum signal value of N first sample
3. Highest signal value is added to variable

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4. Repeat step 2-3 for second N sample until last N sample.

Table 6. Comparation Among Fundamental Frequency

Blade name	Fundamental frequency (Hz) with Accuracy 1Hz		
	GStrings	FFT	AWPM
SaronSlendro1	533	533	533
SaronSlendro2	612	612	612
SaronSlendro3	700	699	699
SaronSlendro5	800	801	801
SaronSlendro6	930	930	930
DemungSlendro1	136	267	266
DemungSlendro2	307	306	306
DemungSlendro3	348	348	348
DemungSlendro5	403	403	404
DemungSlendro6	462	462	462
PekingSlendro1	1079	1079	1079
PekingSlendro2	1239	1239	1239
PekingSlendro3	1420	1420	1419
PekingSlendro5	1651	1651	1651
PekingSlendro6	1930	1931	1930
DemungPelog1	298	298	298
DemungPelog2	321	321	321
DemungPelog3	347	347	347
DemungPelog4	404	404	403
DemungPelog5	432	433	432
DemungPelog6	463	463	463
DemungPelog7	512	511	511
SaronPelog1	599	601	599
SaronPelog2	640	640	640
SaronPelog3	691	690	689
SaronPelog4	800	800	800
SaronPelog5	866	866	866
SaronPelog6	926	926	925
SaronPelog7	1010	1010	1010
PekingPelog1	1221	1223	1222
PekingPelog2	1297	1298	1299
PekingPelog3	1404	1405	1404
PekingPelog4	1653	1654	1653
PekingPelog5	85	1777	1778
PekingPelog6	1932	1932	1932
PekingPelog7	2109	2109	2109

Figure 10 is the envelope result of Demung Slendro 1 and comparison envelope. The black plot is the comparison envelope, while the gray plot is the envelope result of recorded sound.

The more sample value is being used, the smoother and shorter the envelope will be. In contrary, lesser sample value that is being used, the rougher and longer the envelope will be. In other words, higher sample value means more information loss. In this case, the number of information of signal sample which become neglected is higher. Therefore, we need to set an appropriate number of sample value in enveloping process, so that the information in a signal won't lose too much.

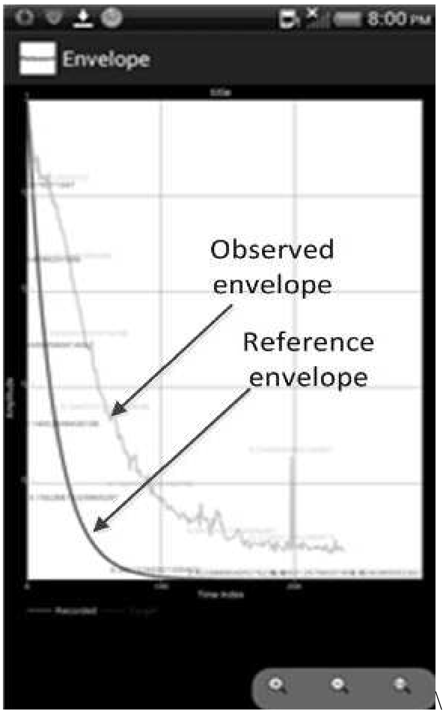

Fig. 10. Result Of Envelope Plot Of Demung Slendro 1 And Comparison Envelope

The formed envelope is then modeled into mathematical equation to ease the plotting process on developed software. This process is done by doing regression. The regression that is implemented has to be adjusted according to the shape of the envelope. Technically, the choosing of regression method is done by observing Sum Squared Error (SSE) value of each regression

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method. The smallest SSE value is indicating that the predicted mathematical model is similar to real model (envelope). SSE is calculated by using Eq. (17).

$$SSE = \sum_{i=0}^{l} \left(m_p(i) - env(i)\right)^2 \quad (17)$$

Where $m_p(n)$ is the predicted model of envelope, l is the sample length of $m_p(i)$ and $env(i)$.

Exponential regression is chosen because it produces smallest SSE compared to another regression model. Table 7 shows the SSE value for each different regression method. Smaller SSE indicating that the predicted mathematical model is similar to real model (envelope).

Table 7. Frequency Estimation RESULTS COMPARISON Using Several Methods

Regression Model	SSE value of each instrument		
	Demung	Saron	Peking
Exponential	124,54	289,61	120,65
Power	11305,53	10930,67	6583,72
Logarithmic	2744,37	1988,23	1400,31

6, CONCLUSION

By implementing this method, we can do the parameters estimation of the signal, simply by using a shorter sample. For example: to estimate the fundamental frequency of Saron Slendro 1, the FFT requires the sample length of 44100 (according to the WAV files used). By using proposed method, it requires only 2000 samples (+/-45 ms) for resolution of 1 Hz, and 9000 samples (+/- 204 ms) for resolution of 0.25 Hz. Moreover, this method does not need to observe the signal at the possibility of signal frequency. This method uses only a certain range, it is not necessary to observe all possible signal frequency.

The comparison envelopes for Demung, Saron and Peking, in form of exponential regression, are as follows: Demung $0.8223e^{-1.194\times10^{-5}n}$, Saron : $0.7693e^{-1.03\times10^{-5}n}$, Peking : $0.7663e^{-1.709\times10^{-5}n}$

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