Filter IIR (Butterworth) and ICA for Identifying Silent Chain's Sound Characteristics

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Abstract: X Company as a silent chain manufacturer has a problem identifying the silent chain's sound characteristics for comparison among other products. Sound characteristics of the silent chain consist of amplitude, frequency, and sound pressure level (SPL). To attain sound characteristics, a filter that can separate the silent chain's audio among other components is needed. Two options that sprout are Butterworth Filter and also ICA. The design of the Butterworth filter is based on identifying the pulse transfer function H(z) that satisfies the requirements of the filter specification. ICA uses mathematical and statistical approaches to decompose components in the observed data set. Singular Value Decomposition (SVD) model used in ICA. In application, sound sources attained from test rig which is consists of a silent chain set (gear, silent chain, and a DC motor). The filter program will be made in Matlab software with a time-domain plot and spectrogram as the outputs. ICA and Butterworth filter can separate silent chain audio. Silent chain's frequencies were ranged from 7000-14000 Hz, and the motor's frequencies are ranged from 0-1000 Hz. As a comparison, the Butterworth filter can work better than ICA because it can minimize noise frequency cleaner and the silent chain's frequency more visible.

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1 INTRODUCTION

PT. X is a manufacturing company that produces Silent chains. In its development, the chain is better than the roller chain. The main advantage of the silent chain is that the sound is quiet and able to operate at a higher speed than the roller chain, which is currently widely used in the industrial as a mechanical power transfer. But, there is no data regarding the characteristics of Silent chain sound, where the data can be used as one of the ingredients for the comparison of competitor products. In addition, the determination of the characteristics of the silent chain is inseparable from the basic reference to be used.

Determination of the sound characteristics required for the silent chain, such as sound, frequency, amplitude and sound pressure level (GOYAL, 2018). To solve the problems, the authors made a test rig design to simulate the sound of a silent chain. But, there is a challenge in determining characteristics of sound (Maulana & Andono, 2016), which is the sound of a silent chain that has mixed up with other components. So, filter the sound is needed to get the actual silent chain's sound characteristics (Hansen, n.d.).

A digital filter IIR (Infinite Impulse Response) with a Butterworth response and ICA (Independent Component Analysis) filter used to analyze the sound. And the A-weighted filter (A-weighted filter) is used to provide a response that has a basic international standard (SI). The filter is used to improve signal quality, such as removing or reducing noise, to retrieve information signals or to separate two or more signals that were previously combined (Lie, 2017).

A MATLAB program is used to process the sound data with the output in the form of sound characteristics of the silent chain (Mathworks, 2008). Filtering with a digital filter IIR using the Butterworth response is best used for audio signals because it has a flat response in the passband and stopband (no ripples). So in its use, this filter is able to produce a better output signal. An ICA filter will be designed to separate the silent chain sound from the motor. Singular Value Decomposition (SVD) method is used as ICA mathematical modeling. Namely determining

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the rotation matrix, stretching matrix, and rotation to minimize the gaussian properties. And the application of A-weighting filter in determining the characteristics of the silent chain sound is inseparable from the sensitivity of the human ear, therefore the output signal from the A-weighting filter has a standard that is in accordance with the international standard.

2 SOUND CHARACTERISTICS

Sound basically a wave. There are two types of wave we commonly know. There are trasnversal, and the other one is longitudinal. Sound is one of longitudinal waves. As a longitudinal waves, there are compressions and rarefactions. Compression is area where you can find dense waves in one time. And Rarefaction is area where you can find tenuous waves in one time. The best three items to describe soundwave are frequency, amplitude, and time-period.

- Frequency : The best thing to describe frequency is number of waves per time. Or commonly, we called as number of waves per second.
- Amplitude : When you draw waves in 2D, you can see normal line, and commonly there are some point that have peak state (in y line). That peak is called amplitude, or we can say the maximum displacement from its normal line. This phenomenon caused by particles of the medium get displaced temporarily from their original undisturbed positions. The International unit of amplitude is metre (m) but sometimes it is also measured in centimetres.
- Time-Period : The best definition to time-period is time required by a wave to complete it's one cycle. In 2D explanation, you can describe it by 'mountain' and 'valley'. Or we called as full vibration. Symbol of time-period is T, and the unit is second (s). (GOYAL, 2018).

3 IIR (INFINITE IMPULSE RESPONSE) FILTER

Infinite Implus Response or IIR is filter in that the output is computed using current and previous input, in addition also the current and previous output. Because of that, this filter is called recursive filter because the output is not straightforward used, but calculated again and again. This filter uses Transfer Function (Hz) that met the requirement of the filter specifications. This method encourage the user to make analogue model, and transform into pulse transfer. Alternately, you can use digital design. (Tutorials, 2018).

Basically transfer function of IIR filter which has n orde is :

$$H(z) = \frac{B(z)}{A(z)} = \frac{b_0 + b_1 z^{-1} \ldots + b_n z^{-n}}{1 + a_1 z^{-1} + \cdots + b_n z^{-n}} \qquad (1)$$

Where :
H(z) = transfer function of IIR filter
a1, a2 = feedback coefficient IIR filter
b1, b2 = feed forward coefficient IIR filter

The frequency response of the Butterworth Filter approximation function is also often referred to as "maximally flat" or when you see in graphs, there are no some kind of noise, or we called it a ripple. So in context of sound, when you want a certain frequency band as your output, then in the result you won't hear some sound from frequency outside your chosen band. (Brown, 2014).

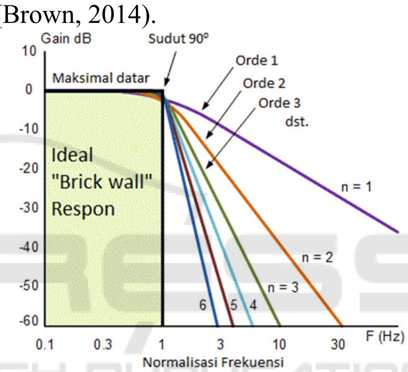

Figure 1: Frequency Response of Butterworth

In this research, author uses the bilinear transform (transformation from continuous-time systems to discrete-time systems (in the Z-domain)). This method has two following characters:

- If H (s) of the laplace transformation is a causal and stable LTI system, then H (z) will be causal and stable.
- The characteristics of H (s) are as initial characteristics of the characteristics of H (z) meaning that in this method the H (s) is needed (Komal Singla, 2014).

4 ICA (INDEPENDENT COMPONENT ANALYSIS) FILTER

Listeners get lots of signal sources every day and every time. Whether it comes from friction (sound waves), or electromagnetic signals that we can not

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catch directly, but we feel (for example through television or radio) (Aapo Hyvärinen, 2000). Listeners deal with these signals every day. However, due to external situations, it is not possible to get signals purely, but signals are mixed (linearly mixed) that we can observe. Because humans get the signal already in mixed form, then this signal is often also called naturally mixed / linearly mixed. Listeners want the original signal for analysis and other purposes. This is the problem of Blind Source Separation, which is to get the original signal from a signal that is naturally mixed / linearly mixed and by observation, indeed only that signal is obtained (Atmaja, Aisyah, & Arifianto, 2010).

To begin the process with ICA, it must start with a simple abstract, namely with the Cocktail Party Problem approach as explained in the previous discussion (Filho, 2012). People who want to use ICA must understand ICA abstractly why ICA should be used. Therefore, with the Cocktail Party Problem approach, people will easily understand that ICA will output one signal source from naturally mixed / linearly mixed signals. Of course, this output does not always produce good quality or perfect output because the method is still being developed and will continue to develop (Kutz, Independent Component Analysis 1, 2018). However, at least with the ICA method, the components of the ICA process will be seen and can be seen as its characteristics even though it is not perfect (Shangmeng He, 2017).

The simple formulation of the object observed with existing components can be written as follows:

$$x = As \tag{2}$$

Where x is the sound that is heard (natural mix), A is the mixing matrix (mixing matrix), and s is a component (sound source) (Kutz, Independent Component Analysis 2, 2018).

The Singular Value Decomposition approach will assume that A is a complex matrix and can be derived into:

$$A = U \sum V^{*} \tag{3}$$

Therefore, the previous equation can be changed to:

$$\vec{X} = (U \sum V^{*})\, \vec{S} \tag{4}$$

Because you want to find the components, then from the formulation above, you must make an inversion, that is:

$$\vec{S} = (U^{*} \Sigma^{-1} V) \vec{X} \tag{5}$$

Where

$$U^{*} = \begin{pmatrix} \cos\theta_0 & \sin\theta_0 \\ -\sin\theta_0 & \cos\theta_0 \end{pmatrix} \tag{6}$$

$$\sum{}^{-1} = \begin{pmatrix} \frac{1}{\sqrt{\rho_1}} & 0 \\ 0 & \frac{1}{\sqrt{\rho_2}} \end{pmatrix} \tag{7}$$

$$V = \begin{pmatrix} \cos\varphi_0 & \sin\varphi_0 \\ -\sin\varphi_0 & \cos\varphi_0 \end{pmatrix} \tag{8}$$

(Kutz, Independent Component Analysis 3, 2018)

5 RESEARCH METHODOLOGY

5.1 Sound Identification

The main approach to this methodology is quantitative method, because we'll play with numbers and data driven. To attain the data, need to identify the sound. Reading data sheets and also study literature about silent chain needed in this step.

5.2 Sound Sampling

The next step is doing some recording to get the sound sampling. In this step, collaborating with mechanical team will be an advantage. An accoustic room is needed to reducing noise, All sound in different environtment were sampled. After that, pull a conclusion to best describe the silent chain characters, such as frequency, amplitude, and period.

5.3 Filter Design

After we get some datas regarding silent chain sound, then it's time to design the filter. For Butterworth (IIR) filter, a table for frequency start and cut-off is needed, and some calculations to gain the amplification of Butterworth filter to get the best results. For ICA Filter, designing will be tricky, need choose the best method to define the random matrix. For this case, the best method is Singular Value Decomposition because it can directly process the sound amplitude into ICA process.

5.4 Transform and Load

The next step is we transform the record data into a mathematical matrices and process it into matlab. 1 file will be processed in ICA filter, and another copy will be processed in Butterwoth filter. The data will transformed (by coding in ICA filter design, or by coding in Butterworth filter design). Finally, load it using graphs and add some insightful information.

6 RESULTS

From the silent chain recordings, amplitude datas can be visualized as a time-domain graph and frequency-domain graph. For a convenient use, a spectrogram can be used to analyze frequency spektrum for each time-domain.

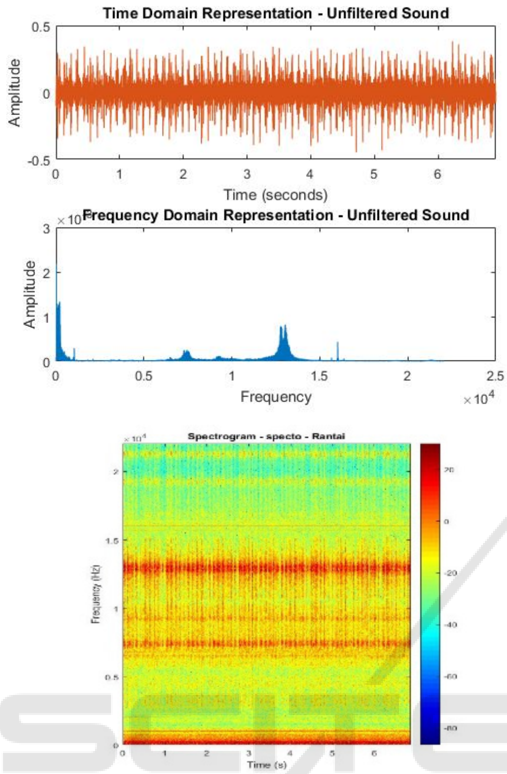

Figure 2: (upper) a time-domain plot from recording data, (center) a frequency-domain plot from recording data, (lower) a spektrogram of frequencies spektrum in time-domain.

That is data from recording section that had been done by microphone + software.

6.1 Butterworth Design and Result

Make a design for identify the filter specification from data attained from figure 2 :

Table 1: Filter Specification

f_{s1}	10 Khz
f_{p1}	12 Khz
f_{p2}	13 Khz
f_{s2}	15 Khz
Fs	44100
K_1 (attenuation)	1 dB
K_2 (attenuation)	-25dB

From the specification, there are two pass band frequencies, fp1 and fp2, and two stop band frequencies, fs1 and fs2. So it is known that the type of filter design is between the band pass filter and the band stop filter. From the value of fp1 and fp2 where the value of fp1 is greater than fs1, it can be concluded that the filter used is a type of band pass filter.

For filtering, we need an orde of the filter. But from the specification, we have to determine the frecuency cut off. So, we got the orde of the filter is :

$$N = \frac{log\left[\frac{\left(10^{\frac{1}{10}}-1\right)}{\left(10^{\frac{25}{10}}-1\right)}\right]}{2log\left(\frac{1}{14.60483064}\right)}, N = \frac{log\left[\frac{(10^{0.1}-1)}{(10^{2.5}-1)}\right]}{2log\left(\frac{1}{14.60483064}\right)}$$

$$N = \frac{log\left[\frac{0.258925412}{315.2278}\right]}{2log\left(\frac{1}{14.60483064}\right)},$$

$$N = \frac{-3.085449789}{-2.32899305} = 1.324799912 \approx 2 \tag{9}$$

Then, do the transfer function H(z) of LPF

$$H(z) = H_{BPF}(S)\Big|_{s=2\left(\frac{1-z^{-1}}{1+z^{-1}}\right)}$$

$$= \frac{0.129837112\left[2\left(\frac{1-z^{-1}}{1+z^{-1}}\right)\right]^2}{\left[2\left(\frac{1-z^{-1}}{1+z^{-1}}\right)\right]^4 + 13.62247648\left[2\left(\frac{1-z^{-1}}{1+z^{-1}}\right)\right]^2 + 46.89311737}$$

$$= \frac{0.519348449(1-z^{-1})^2(1+z^{-1})^2}{16(1-z^{-1})^4 + 54.48990592(1-z^{-1})^2(1+z^{-1}) + 46.89311737}$$

$$= \frac{0.519348449 + 0.519348448\,z^{-2} + 0.519348449z}{117.3830233 - 118.4899059z^{-1} - 108.9798118z + 150.4899059z^{-2} - 9.510094076z^{-3} + 16z^{-4}} \tag{10}$$

Then, do the Y(z)

$$Y(z) = 1z^{-2}\,Y(z) - 1z\,Y(z) + 0.136305912\,Y(z) + 0.004424391\,X(z)$$

$$\begin{aligned}&-0.004383061z^{-1}\,X(z)\\&+\ 0.003451052z^{-2}X(z)\\&-\ 0.054610233z^{-3}\,X(z)\\&+\ 0.032459278z^{-4}\,X(z)\\&+0.004765547z\,X(z)\end{aligned} \tag{11}$$

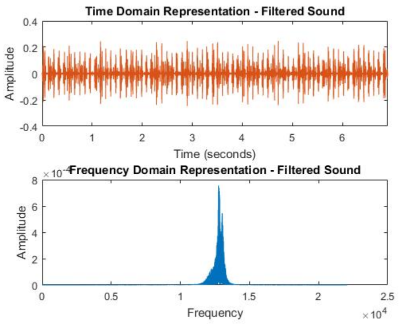

Figure 3: Signal After Filter

From figure 3, the frequency spectrum was more concentrated at one ranged frequency.

6.2 ICA Design and Result

From these measurement data, a filter mathematical model can be made. ICA Filter Mathematical Model is

$$\begin{aligned}&[S1\,S2]\\&=\left(\begin{bmatrix}\cos 21{,}2901830612040^0 & \sin 21{,}2901830612040^0\\ -\sin 21{,}2901830612040^0 & \cos 21{,}2901830612040^0\end{bmatrix}\right.\\&\times\begin{bmatrix}\frac{1}{19{,}2818998848281} & 0\\ 0 & \frac{1}{29{,}0470573960875}\end{bmatrix}\\&\times\left.\begin{bmatrix}\cos 6{,}16147838335633^0 & \sin 6{,}16147838335633^0\\ -\sin 6{,}16147838335633^0 & \cos 6{,}16147838335633^0\end{bmatrix}\right)\\&\times\left(\begin{bmatrix}-0{,}0456943771812110\\ -0{,}0428562424155860\\ \vdots\\ -0{,}0420017502280860\end{bmatrix}\right.\\&\times\left.\begin{bmatrix}0{,}0240345463136426\\ 0{,}0191517338136426\\ \vdots\\ -0{,}0398082271238574\end{bmatrix}\right)\end{aligned} \tag{12}$$

From ICA Filter, the result is two data that must be analyzed, which one that can reflects which component.

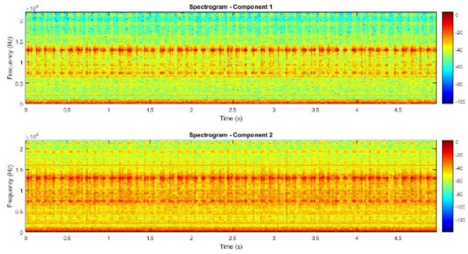

Figure 4: Two Signals After Filter

From figure 4, most likely better signal is the upper one. Because it can focus to silent chain's audio frequencies. But on that graph, we can still see the lower frequency still exist (not completely filtered).

7 CONCLUSIONS

As a final comparison, signal after filter give more insight if we see the Butterworth one. But you must know that to make / design Butterworth Filter, it's not easy, and you must calculate it meticulously. Because, different with ICA which doesn't compel the researcher to have some prior knowledge, Butterworth filter compel the researcher to have knowledge about frequencies, attenuation, and etc.

ICA filter as a new research method to sound separation not give a quite good filtered signal. But the result was quite impressive. Because after design the mathematical model, researcher can just study from one certain component to find out which one the silent chain's frequency, and which one that don't.

As a filter, Butterworth is dependable and reliable. Its method and result is excellent and can make noise frequencies 'disappear' completely. But, researcher must know in the first time, what the silent chain's frequency range is, and must know to what extent the attenuation is permitted.

The final characteristics of silent chain is from 7000 Hz to 14000 Hz. And the motor is 0 Hz – 1000 Hz. Both filter succeed to passing the 7000 Hz – 14000 Hz frequencies.

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