

Performance Enhancement of Whistle Sound Source Tracking Algorithm using Time-Scale Filter Based on Wavelet Transform

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Abstract : A purpose of developing a sound source tracking system in this paper is to reduce the noise efficiently from the received signal by microphone array and measure the signal's time delay between the microphones. I have applied the wavelet analysis algorithm to the system and calculated the sound source's relative position. For the performance evaluation, I have compared with the results of utilizing the digital filtering methods based on the FIR LPF using Kaiser window function and the inverse Chebyshev IIR LPF. As a result, I have confirmed the fact that 'time-scale' filter using inverse discrete wavelet transform was suitable for this system.

Key words : Microphone array, Time delay, Time-scale filter, Kaiser window, Inverse Chebyshev filter, Inverse discrete wavelet transform

1. Introduction

navigators can exchange their brief navigation informations and intentions using whistle system under the restricted visibility condition like a foggy weather. But they have some difficulties in hearing the signal sound on account of a few reasons like the noise around the navigation bridge, navigator's physical condition, etc. So they have a tendency to neglect making use of the ship's whistle system and do not trust the sound information received from other ships. Specially at the vessel which bridge is totally enclosed, the officer on the watch can not even hear outside sound signal. Therefore IMO(International Maritime Organization) has adopted the 'Sound Reception System' as new navigation equipment which can receive the whistle sound signal and indicate the approximate direction of incoming signal(Park, 1992; Koo, 2000).

This study, it is designed to develop a sound source tracking system that is able to measure the range and relative bearing of sound source by utilizing the whistle blast with a higher accuracy than 'Sound Reception System'. Before I reported the algorithm of 2-dimensional sound source's position tracking and the principal for measuring the sound signal's time delay between the microphones(Moon, 1998; Moon, 2000). The physical characteristic of sound signal propagated through sea field and received at the ship has been changed by the reason of the noise produced from a various type of machinery and the phenomenon of sound's reflection and refraction, etc.

In this paper, the time scale filter based on a wavelet

transform algorithm were designed to develop the optimal system which could reduce the noise of sound signal efficiently and find the time delay between each microphones with a high accuracy than the traditional digital filters. These were the finite impulse response type low pass filter using Kaiser window function, infinite impulse response type low pass filter using a chebyshev polynomial. For the performance evaluation, the received data sequences were processed by three filtering methods, the sound source's relative positions were calculated and compared with each other.

2. Wavelet Transform

The Fourier analysis to be used through the whole signal for the analysis(Meyer, 1993; Daubenchies, 1996; Lee, 2002). Especially the wavelet transform can be used to resolve the non-stationary signals both in time-frequency domain simultaneously. And this transform can extract the partial frequency component efficiently through the large scale wavelet applying to the low frequency band and the small scale wavelet to the high frequency band.

Daughter wavelets are obtained from one simple mother wavelet of the changing of scale and translation parameter. These are defined by Eq. (1):(Rioul, 2002)

$$\psi_{b,a}(t) = \frac{1}{\sqrt{a}} \psi\left(\frac{t-b}{a}\right), a, b \in \mathbb{R} \quad (1)$$

where $\psi_{b,a}(t)$ is the daughter wavelet, $\psi(t)$ is the mother wavelet which is called by the prototype function and

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thought to be the basic building block, $a > 0$ is the scale parameter, b is the translation parameter.

The continuous wavelet transform of the time series $f(t)$ is the function $W^f(b, a)$ defined by

$$W^f(b, a) = \frac{1}{\sqrt{a}} \int_{-\infty}^{+\infty} f(t) \psi^* \left(\frac{t-b}{a} \right) dt, \quad (2)$$

where the symbol ψ^* denotes a conjugate form of ψ . Eq. (2) can be interpreted as an inner product of $f(t)$ with the scaled and translated versions of the mother wavelet.

When the wavelet has a form of complex, results of wavelet transform shall be interpreted by a module and a phase. We can regard the module and the phase as a amplitude spectrum and a phase spectrum of the Fourier transform. From this module, we can find how the energy of each frequency component is distributed in the time domain. And the wavelet's scale having maximum value at the module is entirely coincided with a frequency of original signal or similar.

If the wavelet satisfies some admissibility conditions, the inverse wavelet transform can be given by

$$f(t) = \frac{1}{C_g} \int_{-\infty}^{+\infty} \int_{-\infty}^{+\infty} \frac{1}{\sqrt{a}} W_{b,a}^f \psi \left(\frac{t-b}{a} \right) \frac{dadb}{a^2} \quad (3)$$

$$\text{where } C_g = \int_{-\infty}^{+\infty} \frac{|\Psi(a\omega)|^2}{|a|} da < \infty, \quad (4)$$

$\Psi(\omega)$ is the fourier transform of $\psi(t)$.

In this paper, the morlet wavelet was chosen as a mother wavelet. This is a complex wavelet function and have a better capability of separating the contents in the time-frequency domain.

Fig. 1 shows three sound signals which were received by the microphones and collected from AD converter simultaneously. A sampling frequency of the AD converter was 1.0MHz. The amplitude spectral density of the first signal of Fig. 1 is plotted in Fig. 2.

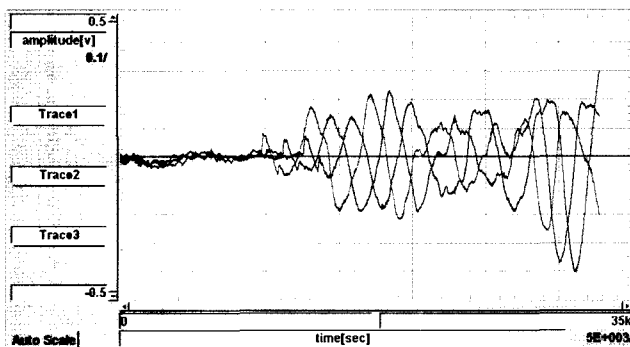


Fig. 1 Received sound signals by microphone sensors

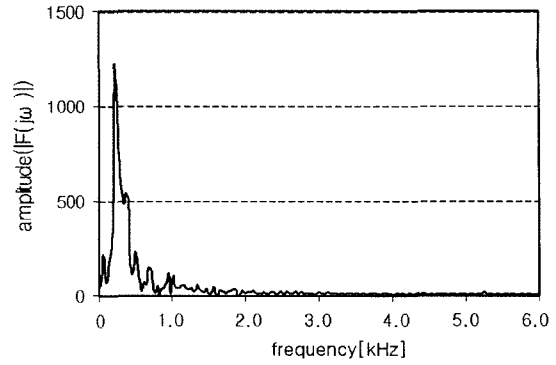


Fig. 2 Amplitude spectrum of the first signal of Fig. 1

Fig. 3 displays the module obtained by the wavelet transform using the morlet wavelet. The scale parameter was ranged from 0.05 to 4.5 and the total number of scales were 100. Dashed rectangular area in Fig. 3 has non-stationary property. Particularly, the rear of the signal have been separated to two different scale bands. Fig. 4 displays the result of inverse discrete wavelet transform of Fig. 3. This is known as the wavelet plan that gives us a very important information about the change of signal's waveform at each scale. And the original time series can be reconstructed by the sum of the real part of the wavelet plan over all scales.

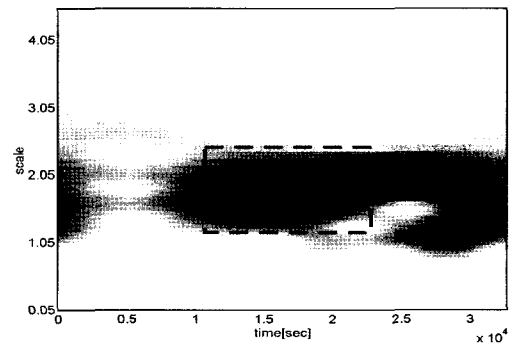


Fig. 3 Module of wavelet

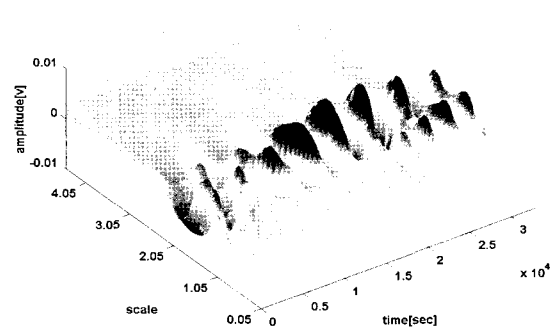


Fig. 4 Wavelet plan obtained by inverse discrete wavelet transform

3. Designing of Digital Filter

3.1 Time-Scale Filter

Time-scale filtering method can be realized by the inner product of the wavelet plan and window function like rectangular. The rectangular window function is given by

$$w_{ij}(b, a) = \begin{cases} 1 & \begin{cases} a_1 \leq i \leq a_2 \\ b_1 \leq j \leq b_2 \end{cases} \\ 0 & \text{(otherwise)} \end{cases} \quad (5)$$

where a_1 and a_2 are the lower limit and the upper limit of scale domain, b_1 and b_2 are the lower limit and the upper limit of time domain.

The time-scale filtering function is defined as

$$f_{ij}(t) = \sum_{a=a_1}^{a_2} w_{ij}(b, a) \cdot W'(b, a)^{-1} \quad (6)$$

Eq. (6) means that the original time series can be filtered in the time-frequency domain at the same time.

Fig. 5 shows the filtered signals of Fig. 1 by the time-scale filter. From this figure, we can know that the rear part signal corrupted by some reasons like noise could be reconstructed to the original frequency components precisely.

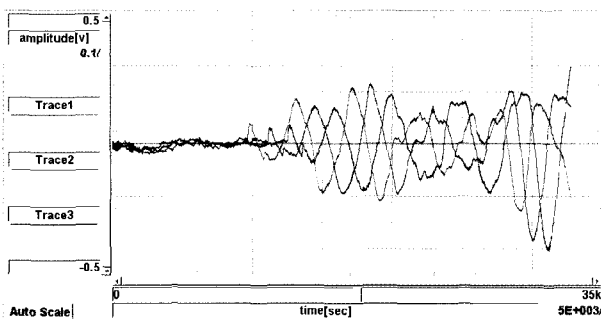


Fig. 5 Filtered sound signal using time-scale filter

3.2 FIR and IIR Filter

In this paper, two digital filters were designed for validating whether the time-scale filtering could obtain more accurate time delay than the traditional filtering methods. These filters are the FIR Low Pass filter and the IIR Low Pass filter. In general, FIR filter can have an exactly linear phase response in time domain. This implicates that the signal can be processed and filtered without any phase distortion. But FIR filter requires more coefficients for sharp cut-off response characteristic than IIR filter. IIR filter can easily realize the sharp cut off response characteristic without increasing the number of coefficients(Ifeachor, 1993).

At first, Kaiser window function was adopted as a basic function of FIR filter, because this function has a ripple control parameter that allows us to design the filter specification like transition width and stopband attenuation without any restriction. The Kaiser window is given by

$$w(n) = \frac{I_0\left\{\beta\left[1 - \left(\frac{2n}{N-1}\right)^2\right]^{\frac{1}{2}}\right\}}{I_0(\beta)} \quad \text{otherwise} \quad (7)$$

$$-\frac{(N-1)}{2} \leq n \leq \frac{(N-1)}{2} = 0$$

where $I_0(x)$ is the zero order modified Bessel function of the first kind. β controls the way the window function tapers at the edges in the time domain.

Fig. 6 shows the frequency response of FIR LPF using Kaiser window function. This filter was designed to satisfy the following specifications:

passband edge frequency	0.25kHz
passband ripple	1.0dB
stopband edge frequency	2.1kHz
stopband attenuation	25.0dB

And the number of filter coefficients was 191.

IIR LPF was designed using inverse Chebyshev type that had a characteristic of equal ripple in the stopband and monotonic in the passband. This filter could be realized the sharp cut-off by a small number of filter coefficients. Inverse Chebyshev filter can be characterized by the magnitude-squared response

$$|H_{CN}(j\omega)|^2 = \frac{\varepsilon^2 C_N^2(1/\omega)}{1 + \varepsilon^2 C_N^2(1/\omega)} \quad (8)$$

where $C_N(\omega)$ is a Chebyshev polynomial which shows the equal ripple in the passband, N is an order of the polynomial, ε means the passband ripple, and ω is the angular frequency.

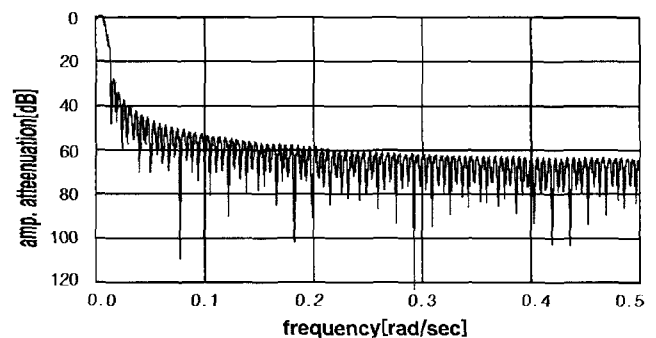


Fig. 6 Frequency response of FIR LPF(normalized frequency scale)

Fig. 7 shows the frequency response of IIR LPF and this filter was designed to satisfy the following specifications:

passband edge frequency	0.25kHz
passband ripple	0.1dB
stopband edge frequency	0.65kHz
stopband attenuation	40.0dB

And the order of filter coefficients was 4.

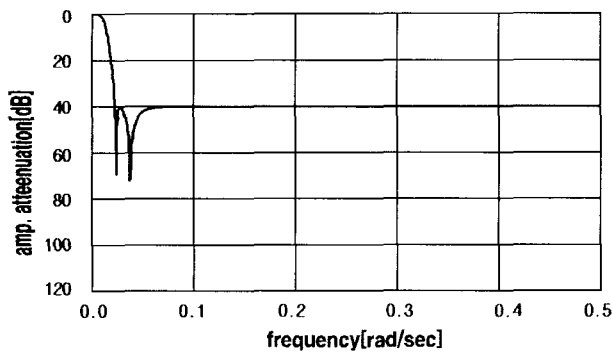


Fig. 7 Frequency response of IIR LPF(normalized frequency scale)

Fig. 8 and Fig. 9 show the filtered signals by FIR LPF and IIR LPF. In the case of FIR LPF, the wave correspond to the low frequency was almost sustained, so there was no remarkable filtering effect. But IIR LPF introduced better smoothing and eliminating performance than FIR LPF.

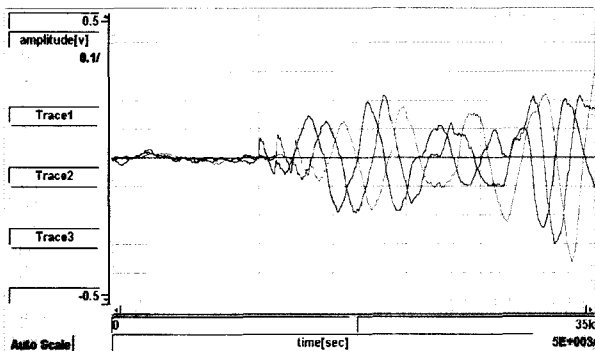


Fig. 8 Filtered signals by FIR LPF

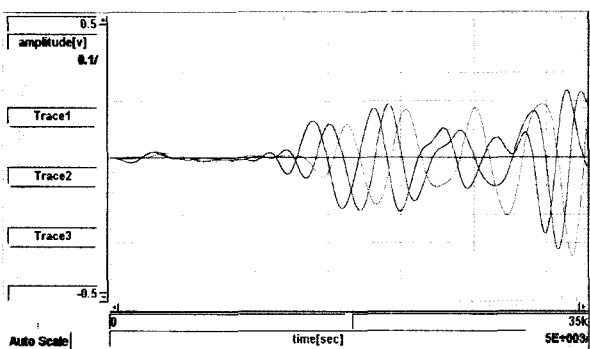


Fig. 9 Filtered signals by IIR LPF

4. Experiments and Evaluations

4.1 Experimental Environment

Several experiments were conducted in a gymnasium, length about 30meters, width about 15meters, height about 10meters. Three condenser microphones were arrayed at 0.5meter intervals in a straight line(Moon, 2000). The distance from the sound source to the center microphone of array was set to two cases, 9.53meters and 11.75meters. And sound source was arranged within the $-90^{\circ} \sim +90^{\circ}$

circle randomly. The sound signals were measured 15 times continuously at the same position of source.

Fig. 10 shows the sequence of signal processing and the whistle sound source tracking system. The received signals were converted to digital data and processed using three filtering methods. After measuring the time delay from the filtered data, positions of sound source were calculated and analyzed.

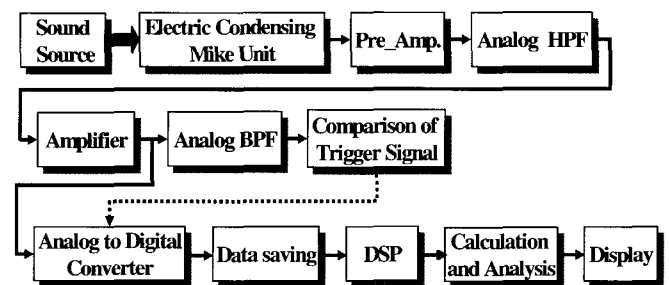
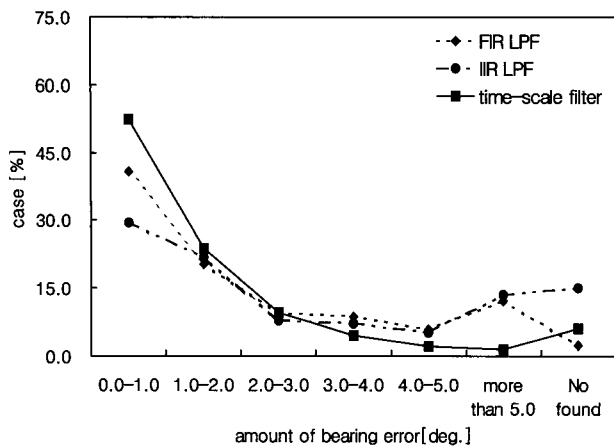


Fig. 10 Diagram of signal processing

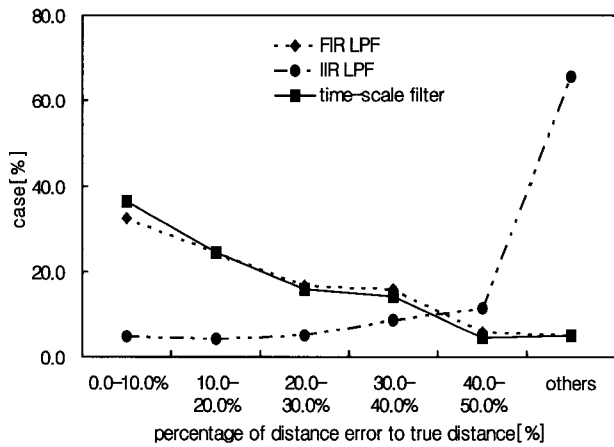
4.2 Result of Experiments

4.2.1 Bearing Measurement

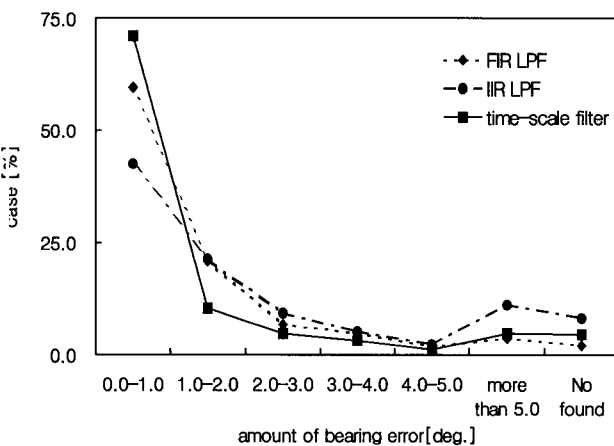
To confirm the accuracy of relative bearing measurement according to each signal processing methods, amount of measured bearing error was obtained by the comparison the measured relative bearing with the true value. And the measurement ratios were calculated according to the each amount of bearing error. Fig. 11 shows the analysis of this sound source's relative bearing measurement. The horizontal axis indicates the amount of bearing error and the vertical axis is the percentage of measured cases within amount of bearing error. In case of measured bearing error under 1.0° , the time-scale filtering method recorded about 10%~20% higher than the FIR LPF and the IIR LPF filtering method. And I have found that the time-scale filtering method could measure the sound source's bearing with higher accuracy than any other methods.



(a) distance 9.53m



(a) distance 9.53m



(b) distance 11.75m

Fig. 11 Result of bearing measurement according to each filtering method

4.2.2 Distance Measurement

Fig. 12 shows the result of the distance measurement. Analysis was conducted by the similar process with the bearing measurement analysis. In this figure, the horizontal

axis indicates a percentage of the distance error according to the true distance. And the vertical axis is the percentage of measured cases within some extent of distance error.

In the experiments of distance measurement, the received sound signal data within $-40^\circ \sim +40^\circ$ were selected. When the sound sources are located outside of this scope, there is little or no time delay between the microphones. So the distance measurements are impossible or have too much measuring error. This aspect had been confirmed by the computer simulation(Moon, 1996). Taken as a whole, it was not a satisfactory performance of distance measurement in comparison with the case of bearing measurement. Specially the filtering method using IIR LPF was not suitable for the sound source's distance measurement, because there was 66% of cases which produced a large error or could not calculate the distance. And the time scale filter and FIR LPF showed the similar distance measurement ratios all over the distance error ratios, but the time-scale filtering method could measure about 5% higher than FIR LPF in case of measured distance error ratios less than 10%(95.3cm in Fig.11 and 117.5cm in Fig.12).

The AD converter's sampling frequency of this whistle sound source tracking system was set at 1.0MHz for obtaining better resolution of measurement. And the center frequency of whistle sound signal is always located within the band of 70-700Hz. These environments introduced that FIR filter didn't make a sufficient attenuation at stopband and could not reduce the noise included in the sound signal efficiently. Therefore even though the time-scale filter and FIR filter could realize the linear phase response characteristic, the time-scale filtering method could measure the time delay and the relative position of sound source accurately. And also the case of IIR filter could reduce the noise and make a smoothing of signal sufficiently, but the non-linear phase response characteristic brought about much time delay error in measurement rather than the time scale filter.

As a result, I have confirmed the fact that the time-scale filtering method could detect the time delay and measure the sound source's relative positions with high accuracy in comparison to the others, because it had the linear phase response and sharp cutoff response characteristic simultaneously.

5. Conclusion

In this paper, the time-scale filtering method using the discrete inverse wavelet transform algorithm was applied to

the signal processing part for the purpose of reducing the noise concluded in a sound signal efficiently and finding the time delay accurately. And several experiments were carried out for verifying the effectiveness of the time-scale filter.

As a result, I have found that the accuracy of measurements were differentiated by the methods what kind of filter were adopted. And I have confirmed the facts that the digital signal processing method using time-scale filter could enhance the performance of detecting the time delay between microphones and the positioning of whistle sound source.

References

- [1] Daubenchies, I.(1996), Where Do Wavelet Come From?, Proceedings of the IEEE, Vol. 84, No. 4, pp.510-513.
- [2] Ifeachor E. C., Jervis, W. J.(1993), Digital signal processing, Addison-Wesley, pp. 278-490.
- [3] Koo, J. Y.(2000), International Trends of New Aids to Navigation and IMO Carriage Requirements, Journal of the Korean Institute of Navigation, Vol. 24, No. 4, pp.319-329.
- [4] Lee, S. H., Yoon, D. H.(2002), Introduction to the Wavelet Transform, Jin-Han, pp.8-13.
- [5] Meyer, Y.(1993), Wavelet, Application and Algorithms, Siam.
- [6] Moon, S. B., Jun, S. H.(1996), A Study on 2-dimensional Sound Source Tracking System, Journal of the Korean Institute of Navigation, Vol. 20, No. 4., pp. 71-80.
- [7] Moon, S. B., Jun, S. H.(1998), A Study on 2-dimensional Sound Source Tracking System II, Journal of the Korean Institute of Navigation, Vol. 22, No. 4., pp. 31-36.
- [8] Moon, S. B., Jun, S. H.(2000), A Study on 2-dimensional Sound Source Tracking System III, Journal of the Korean Institute of Navigation, Vol. 24, No. 5., pp. 443-450.
- [9] Park, Y. S.(1992), Marine Traffic Laws, Hyung-Seul, pp.695-735.
- [10] Rioul, O., Vetterli, M.(1996), Wavelet and Signal Processing, IEEE SP Magazine, Vol.8, No.4, pp.14-38.

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