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# A Time-Series Data Analyzing System Using a New Time–Frequency Transform

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## Abstract

*A new time-frequency transform, whose resulting spectrogram has the best frequency resolution among that of existing transforms, is employed to construct a simple system of data acquisition and analysis. The transform employs an iterative and diffusive filter to remove the non-sinusoidal part. The spectrum of the sinusoidal part is evaluated by a strategy involving a Fast Fourier Transform (FFT). By imposing a Gaussian window centered at a given frequency on the spectrum, a band-passed data is obtained which is just the corresponding mode or real part of the transform. Finally, the corresponding amplitude is obtained via the Hilbert transform. A commercial microphone plugged into a notebook computer properly equipped with necessary software and the transform form a complete portable system. Two tests show that the system is a new tool to look into detailed information of a complicated data string formed from a single or multiple sources.*

**Keywords**: *time series data, time-frequency transform, portable system.*

## 1. Introduction

For time series data, as noted by Farge [1], a time frequency analyzing tool can capture the variation of spectrum with respect to time which reflects the involved details and mechanism(s). Therefore, spectrograms generated by time frequency transforms, such as short time Fourier, Gabor and wavelet transforms, are widely applied [2]. Although distributions of original Wigner-Ville distribution and Cohen class are also successful transforms [2,3], the related methodology will go beyond the scope of this study and it is not discussed here.

In several previous studies [4-7], it is seen that both Gabor and continuous wavelet (Morlet) transforms give detailed information by embedding a window to weight the data string on the time domain. These studies also show that a window on the time domain results in a band-pass filtered spectrum. Based on this fact, a new time frequency transformation was proposed in Ref.[8]. In this study, further explanation and application of this transform are shown.

## 2. Theoretical Development

The transform proposed in Ref.[8] is a procedure rather than a single mathematical formula. Under the assumption that time series data has no discontinuous component, the procedure involves the following steps.

1. Apply the iterative filter of Ref.[9] to remove the data's non-sinusoidal part whose contribution to the spectrum is also named as the Direct Current (DC) contamination.
2. Employ a strategy of FFT to generate a spectrum with small error [10].
3. Add Gaussian (or other) window to get a band-pass limited spectrum for a given frequency. The corresponding mode or real part of transform is obtained by applying an inverse FFT to the spectrum.
4. Use the Hilbert transform to evaluate the corresponding amplitude.
5. Plot the two-dimensional spectrogram.

For the sake of completeness, the theoretical content of Ref.[8] is briefly restated below.

### 2.1. Iterative Filter with Diffusive Property [9]

Consider a data string, $y_{0,}y_1,...,y_N$ and expand it as follows.

$$y(t) = \sum_{n=0}^{N-1}(b_n - jc_n)\exp[j2\pi t/\lambda_n] \qquad (1)$$

If the Gaussian smoothing method is employed to smooth the data, it can be shown numerically that it is a diffusive smoothing. But the transition zone of this low-pass filter is too wide. In order to narrow down the transition zone, an iterative filter based on Gaussian smoothing was developed in Ref.[9]. The iteration procedure smoothes the remaining high frequency part repeatedly. If the iteration stops at the $m$ – th step, the final high frequency part $y'(t)$ is the desired short wave part. The final smooth part is

$$\bar{y}(t) = y(t) - y'(t)$$
$$\approx \sum_{n=0}^{N-1}[1 - A_n]^m (b_n - jc_n)\exp[j2\pi t/\lambda_n] \quad (2).$$
$$0 \le A_n(\sigma/\lambda_n) \approx \exp[-2\pi^2\sigma^2/\lambda_n^2] \le 1$$

Like the Gausssian smoothing method, the iterative filter is also diffusive. Supposing a data string has a frequency gap in the range of $\bar{\lambda}_0 < \lambda < \bar{\lambda}_1$ within which all modes are not important, both $m$ and $\sigma$ can be solved by the simultaneous equations with factor $[1 - A(\sigma/\bar{\lambda})]^m$ equal to almost 0 and 1, respectively. If there is no such gap, the above procedure can be considered as initial work. By extracting the data associated with those modes with $\lambda < \bar{\lambda}_1$ from the spectrum of the remaining high frequency part, a sharp filter cutting at $\lambda_1$ is obtained [11].

## 2.2. Spectrum with small error

In Ref.[4-8,10-11], the above mentioned iterative filter is employed to remove the non-sinusoidal and low frequency parts. For the remaining high frequency part, the simple strategy of FFT [10] generates a spectrum with very small spectrum and DC errors. The strategy includes: find zeros at two ends; remove data beyond zeros; redistribute data so that the total number of points is equal to an integer power of 2; do an odd function mapping to ensure periodicity; follow, finally, with an FFT algorithm.

## 2.3. New Time Frequency Transform [8]

Assume that the high frequency part, $y'(t)$ , is expressed in the form of Eq.(1). Any time-frequency transform [1-8] can be applied to generate a spectrogram without DC error. For example, the Gabor transform gives [4]

$$G(f,\tau) = 1/\sqrt{a}\int_{-\infty}^{\infty} y'(t)e^{-2j\pi f(t-\tau)}e^{-(t-\tau)^2/(2a^2)}dt$$
$$\approx \sqrt{\pi a/2}\sum_{n=0}^{\infty}(b_n - jc_n)e^{2j\pi f_n\tau}e^{-4\pi^2a^2(f_n - f)} \quad (3)$$

where $a$ is the scale function. The Gaussian window imposed on the time domain clearly results in imposing a corresponding Gaussian window on the spectrum domain. The Morlet transform and all enhanced Gabor and Morlet transforms have similar mappings of windows between time and frequency domains [5-8].

Based on this fact, a Gaussian window is imposed on the spectrum of $y'(t)$ so that a band-passed data string corresponding to $f_k$ can be obtained [8]

$$y_k(\tau) = \sum_{n=0}^{N-1}(b_n - jc_n)e^{j2\pi\tau/\lambda_n}e^{-(n-k)^2/(2c^2)} \quad (3)$$

where the last exponential term is the Gaussian window with window size $c$. Now $y_k(t)$ is considered as the mode or real part corresponding to $f_n$ . The amplitude of $y_k(t)$ is evaluated either by the original [9] or modified Hilbert transform [8]. The Hilbert transform is fast but has the penalty of convolution error due to end effect. When the modified Hilbert transform is employed, the spectrum, by the proper feeding of 0's to the function $e^{j2\pi f_n t}/t$, is unavailable now so that it should be done on time domain. Finally, the spectrogram is obtained by scanning all frequencies.

Along a $f = f_k$ line, the real part of the spectrogram is a mode with a Gaussain window size characterized by parameter $c$ on spectrum. For the problems of turbulent flow data, modes generated by different values of $c$ correspond to different physical meanings and should be carefully addressed.

## 3. Results and Discussions

The new time-frequency transform together with the proposed mode decomposition can be integrated into a compact program. In this study, a prototype of a portable system is examined. The system employs a commercial microphone for personal and notebook computers to collect speech or acoustic data. Its specifications are 20-20,000 Hz, 100mw, 32 Ω 105db sound pressure level sensitivity at 1kZ ±2 % and uses a 3.5mm stereo jack plug for connection. The data conversion employs the built-in 16 bit recording software of the Micro-Soft Windows XP system.

Figure 1 shows the data and spectrum of a vocalization of "hello", respectively. In Figs.2a-2c, spectrograms generated by the original Gabor, enhanced Gabor, and the new transform are shown, respectively, where 120 uniform spacing is employed to resolve 50 to 1300 Hz and parameter $c$ takes a value of 1. Figure 2a generated by the original Gabor transform attains its best resolution with $a$ at a value of 0.02 sec. The result of the enhanced Gabor transform (Fig.2b), which uses $y_k(t)$ to replace $y'(t)$

in Eq.(3), has a better resolution than that of Fig.2a. The resulting resolution with respect to $a$ is insensitive over the range from 0.02 to 0.03 seconds.

The result of the proposed transform (Fig.2c) attains the best resolution as shown. Although the main feature can be captured from spectrogram of Fig.2a, the three-dimensional plot shows that most of the detailed information is blurry. Comparing to Fig.2a, the enhanced Gabor transform gives a much better amplitude resolution but is still too smeared compared to Fig.2c. This test case shows that the new transform does capture all the detailed information and can be employed to examine insights from the voice reflecting the speaker's character, health condition, mood, and many others.

The second test is a home electrical fan with and without a clump of paper in a flattened shape attached to one of the fan blade (Fig.3). The microphone head is placed at the axial line 20 cm from the axis hub cover. Figure.4 shows the raw data (ranging from 2.5 to 3 sec.) and corresponding spectrum. After employing the proposed time frequency transform, the resulting amplitude plot is shown in Fig.5, where 100 uniform spacings from 2 to 602 Hz are calculated with $c = 1$. In the spectrogram, there are two modes close to each other that form a beat around 60-70 Hz as shown. The high harmonic modes of 60 Hz, say 120 and 240 Hz, persist very well, and those of 180, 300, and 360 Hz do not continuously exist. The harmonic modes (140 and 280 Hz) of the 70 Hz mode do not persist. This reflects that the 70 Hz mode is the blade mode while the 60Hz mode is the mode of the entire structure including the motor and fan. From Fig.4, one can not know what happens with the 15 and 30Hz modes because many modes cluster there. From the time frequency plot, it clearly shows an unsteady low frequency fluctuation which reflects the vibrations of the whole system. On the high frequency region around 500 Hz, there are unsteady vibrations which generate noise. From the spectrogram, it seems that the quality of the fan is not good because of the low frequency vibration and noise.

After attaching a clump of paper to one of fan blades, the raw data and spectrum of Fig.6 show that the undesired paper amplifies the dominate mode (70 Hz) of the fan and generates clear high harmonic modes (modes of 140 and 210 Hz). Because the air flow stream introduces non-linear damping effect, the 280 and 350 Hz modes are not obvious. Dominant and high harmonic modes of the whole structure, say 60, 120, and 240 Hz, are amplified. The discontinuous 180 and 300 Hz modes are also amplified. As compared with Fig.5, the 90, 330, and 420 Hz modes are new. They are the vibration of the whole system and are generated by the heavy vibration of the fan. The unsteady vibration exists from low to high frequency modes and is much more serious than that of Fig.5. This test shows that the proposed simple system can easily grasp many important features of a product.

Many tests reflect that this new spectrogram has intersection problems whenever two modes, which belong to different sources and are mutually out of phase, intersect at a point on the spectrogram plane. Experiences show that many distributions of the Cohen class [3] may partially solve this problem due to dominant modes but do not work for that of minor modes. For a given single mode, the frequency resolution bandwidth of a spectrogram is always thicker than that of a Wigner-Ville distribution. Therefore, the Cohen class can be a complementary tool of the present spectrogram to treat a data string coming from multiple sources. The above discussions indicate that the new transform reflects many insights of a complicated data string. Developments of similar systems using the strain gauge, piezoelectric sensor, photo-sensor, thermo-couple, …, etc. are on the way.

## 4. Conclusions

A portable system consisting of a new time-frequency transform, a commercial microphone and a note book computer is developed. It decomposes complicated time series data either of a human voice or of an engineering system's acoustic radiation into a two-dimensional time varying spectrum. Such a system can be easily constituted and applies to many fields with regard to complicated time series data.

## 5. Acknowledgement

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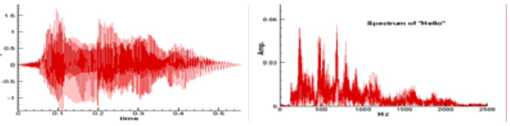
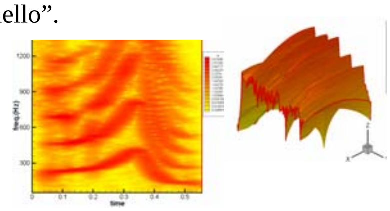
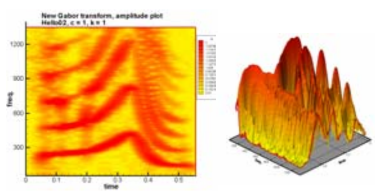


Fig.1 The raw data (left) and spectrum (right) of "hello".

(2a)



(2b)



(2c)

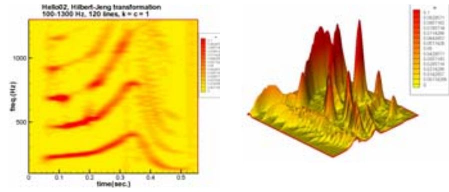


Fig.2 Two and three dimension amplitude plots of the vocalization "hello": (a) the Gabor transform with $a$ =0.02 sec.; (b) the enhanced Gabor transform where $a$ can be 0.002 to 0.03 sec., $c=1$; and (c) the proposed transform with $c=1$.

Fig.3 The photo of a home electric fan with a clump of paper attached to one of the fan blades.

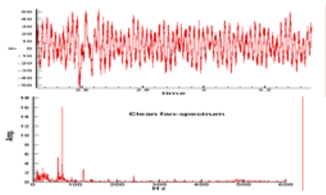


Fig.4 The raw data to be analyzed and corresponding spectrum of the clean fan.

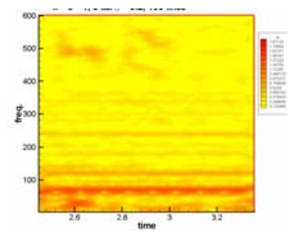


Fig.5 The time frequency plot of the clean fan.

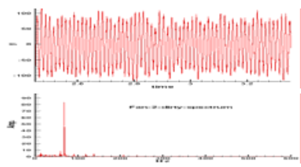


Fig.6 The raw data and corresponding spectrum of the fan with a clump of paper attached .

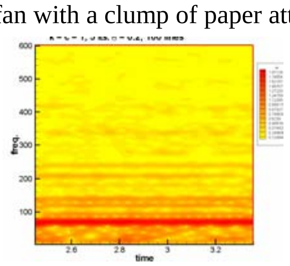


Fig.7 The time frequency plot of the fan with a clump of paper attached to one of the blades.