Noise Removing for Time-Variant Vocal Signal by Generalized Modulation

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The 5th Asia-Pacific Signal and Information Processing Association Annual Summit and Conference October 30, 2013

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- Noise removal: An essential issue for vocal signal processing.
- Well-known techniques: Spectral subtraction methods, Wiener filtering, etc.
- However, **none of** these methods consider that the spectrum of a vocal signal may vary with time.
- Neither do they utilize **the time and the frequency domain** characteristics **simultaneously** to design the filter.

• In order to take into account the varying instantaneous frequency, we exploit the **time-frequency analysis** by using the **short time Fourier transform(STFT)**:



Figure. The FFT of a human vocal signal



Figure. The STFT of the same signal

- Basically, a simple way to remove noise is by using frequency selective filters; e.g., a lowpass filter. However, a great deal of noise might still remain if the signal bandwidth is large.
- Therefore, we try to find out the method to further reduce noise by modifying the signal into one with a **narrower bandwidth** before going to the lowpass filtering step.

• The flowchart of our noise-removing algorithm.



Figure. Flowchart of the proposed noise-removing algorithm.

2. Conventional Modulation Method

Analytic signal generation

- the analytic signal generation is to halve the signal bandwidth
- the negative part of a signal will be removed and the positive part will remain



2. Conventional Modulation Method

Conventional modulation

- a modulation function $m_c(t)$ is an exponential function with the first order phase:

$$m_c(t) = \exp(-j2\pi f_1 t)$$

- by **multiplying** $m_c(t)$, the signal will be frequency-shifted by f_1 Hz in the frequency domain.



2. Conventional Modulation Method

• Apply to the noisy signal



3. Proposed Generalized Modulation

Generalized modulation

- Instead of multiplying a first order exponential function, the modulation function of generalized modulation $m_g(t)$ is an exponential function with a higher order polynomial:

$$m_g(t) = \exp[-j2\pi(a_nt^n + a_{n-1}t^{n-1} + \dots + a_1t + a_0)]$$

The polynomial is derived by approaching the central frequency of the signal



Blue line: the central frequency of the signalBlack line: the approaching polynomial

3. Proposed Generalized Modulation

Generalized modulation

- by **multiplying** $m_g(t)$, the instantaneous frequency of the signal will be shifted by $f_{in}(t)$ Hz, where

$$f_{in}(t) = \frac{1}{2\pi} \frac{d}{dt} \arg\left(m_g(t)\right) = \frac{d}{dt} \left[-(a_n t^n + a_{n-1} t^{n-1} + \dots + a_1 t + a_0)\right]$$
$$= -\left[na_n t^{n-1} + (n-1)a_{n-1} t^{n-2} + \dots + a_1\right]$$



3. Proposed Generalized Modulation

• Apply to the noisy signal



• Fractional Fourier Transform (FRFT)

- we find that, combining the proposed generalized modulation with the FRFT, a very narrow signal bandwidth can be achieved.
- The FRFT assists the generalized modulation in reducing the signal bandwidth by "rotating" the time-frequency distribution in advance.



• Fractional Fourier Transform (FRFT)

- Note that, for the human vocal signal, it has very large negative slope around 0.2 second, which causes the modulated signal to be out-offlatness around 0.2 second. However, if we rotate the time-frequency distribution of the signal in advance, we can alleviate the effect of large slope on the modulated signal



• Apply to the noisy signal



The proposed noise-removing algorithm



- (b) Analytic signal conversion + conventional
- (c) Analytic signal conversion + FRFT + conventional modulation.
- (d) Analytic signal conversion + FRFT + proposed generalized modulation.

- We totally experiment on 4 different time-variant audio signals: 3 human vocal signals and 1 whale voice signal.
- We add the AWGN with different average powers to the signals to see the performances of different noise-removing schemes under various noise conditions.
- The noise reduction performances are measured by the NMSE (Normalized Mean Square Error).

• In each experiment, we compare the following 5 different noise-removing schemes:

1. Original Error

-the noisy signal without any modifications

2. LPF:

-the noisy signal after a proper lowpass filter

3. Analytic + Conventional Modulation + LPF

-the noisy signal modified by the noise-removing scheme based on the conventional modulation

4. Analytic + Proposed Generalized Modulation + LPF

- -the noisy signal modified by the noise-removing algorithm based on the proposed generalized modulation
- 5. Analytic + FRFT + Proposed Generalized Modulation + LPF (Proposed Algorithm)

-the noisy signal modified by the proposed noise-removing algorithm

-The human vocal signal-

- 1. Original Error
- 2. LPF
- 3. Analytic + Conventional Modulation + LPF
- 4. Analytic + Proposed Generalized Modulation + LPF
- 5. Analytic + FRFT + Proposed Generalized Modulation + LPF (Proposed Algorithm)





Figure. Comparison among the performances of different noise-removing schemes on the human vocal signal.

-Human Vocal Signal 2-

- 1. Original Error
- 2. LPF
- 3. Analytic + Conventional Modulation + LPF
- 4. Analytic + Proposed Generalized Modulation + LPF
- 5. Analytic + FRFT + Proposed Generalized Modulation + LPF (Proposed Algorithm)





Figure. Comparison among the performances of different noise-removing schemes on Human Vocal Signal 2.

-Human Vocal Signal 3-

- 1. Original Error
- 2. LPF
- 3. Analytic + Conventional Modulation + LPF
- 4. Analytic + Proposed Generalized Modulation + LPF
- 5. Analytic + FRFT + Proposed Generalized Modulation + LPF (Proposed Algorithm)





Figure. Comparison among the performances of different noise-removing schemes on Human Vocal Signal 3.

-A whale voice signal-

- 1. Original Error
- 2. LPF
- 3. Analytic + Conventional Modulation + LPF
- 4. Analytic + Proposed Generalized Modulation + LPF
- 5. Analytic + FRFT + Proposed Generalized Modulation + LPF (Proposed Algorithm)





Figure. Comparison among the performances of different noise-removing schemes on a whale voice signal.

6. Conclusions

- 1. A new noise-removing algorithm is proposed, which is the combination of the STFT, analytic signal conversion, the FRFT filter, and the proposed **generalized modulation** operation.
- 2. With the proposed algorithm, the area of the signal spectrogram is **reshaped** and the **bandwidth** of the signal is further **minimized**.
- 3. The signal part and the noise part of a time-variant signal are then well separated and a better noise-reducing performance can be achieved.

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Thanks for your attention.