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Modified Algorithm Based on Quadratic Correlation Time Delay Estimation

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Abstract: In the passive positioning system, the traditional generalized cross-correlation algorithm estimates the effect under the influence of noise and reverberation, while the delay estimation algorithm after homomorphic filtering reduces the anti-reverb ability of the cross-correlation algorithm, but also reduces the cross-correlation of the signal, resulting in the decrease in the noise resistance of the cross-correlation algorithm. Based on this, the quadratic cross-correlation operation of the full-pass component of the signal is performed after homomorphic filtering to improve its noise immunity. In addition, the Introduction of Hilbert Difference sharpens the secondary cross-correlation peaks to make peak detection that reflects the time delay more accurate. Experimental simulation shows that the proposed method can effectively suppress the influence of noise and reverberation in non-stationary speech signals, and improve the performance of delay estimation.

Keywords: Time delay estimation ;Reverberation; Homomorphic filter; Quadratic cross-correlation; Hilbert

1. INTRODUCTION

Time-delay estimation (TDE) is a fundamental method for identifying, locating, and tracking radiation sources, with the goal of measuring the relative arrival time difference (TDOA) between different channels. Recently, there have been more and more smart devices and applications using voice-based locators. In a room, the sensor receives not only signals from direct paths, but also signals after attenuation and delay of source signals absorbed through reflections from the walls of the room. This multipath propagation effect introduces echo and spectral distortion in the observation signal, called reverberation, which severely reduces the performance of the delay estimation algorithm.

Mainstream methods for delay estimation include adaptive estimation algorithms[1], methods based on higher-order statistics[2], and methods based on cross-correlation [3]. Adaptive algorithms for latency estimation are able to track real-time latency between signals, but estimation accuracy is often limited at low signal-to-noise ratios. In order to eliminate Gaussian noise and improve the estimation accuracy, an estimation method based on a high-order statistic was studied, but the computational complexity was too high. As the most commonly used method, cross-correlation has been deeply studied for its simplicity and good performance. In order to obtain a higher time resolution at a very low signal-to-noise ratio, many improved algorithms based on intercorrelation are proposed. The quadratic correlation algorithm is one of the effective methods to suppress quadratic cross-correlation noise. In this paper, the TDE problem is studied, focusing on the problems of homomorphic filtering against reverberation and Hilbert differential noise immunity.

2. SIGNAL MODEL

In the process of TDOA positioning, the signals emitted by the source are received by sensors with different positions for delay estimation. Take, for example, the continuous signals received by receiver 1 and receiver 2. The continuous signal is set to $x_1(t)$ and the discrete signal is set to $x_2(t)$ and the discrete signal is set to $x(k)$. τ represents the time delay between the signal reaching receiving station 1 and receiving station 2, D represents the number of sample points of the delay, a_1 and a_2 represent the amplitudes of $x_1(t)$ and $x_2(t)$, and $w_1(t)$ and $w_2(t)$ represent Gaussian white noise. The above signals are all smooth Gaussian processes and independent of each other, and the mathematical model of the continuous signal can be expressed as equation (2-1), and the mathematical model of the discretized signal can be expressed as formula (2-2).

$$\begin{cases} x_1(t) = a_1 s(t) + w_1(t) \\ x_2(t) = a_2 s(t-\tau) + w_2(t) \end{cases} \tag{2-1}$$

$$\begin{cases} x_1(k) = a_1 s(k) + w_1(k) \\ x_2(k) = a_2 s(k-\mathrm{D}) + w_2(k) \end{cases} \tag{2-2}$$

where $s(t)$ is the signal sent by the source, $w_1(t)$ is the noise signal during transmission, $x_1(t)$ is the signal received by the monitoring station 1, and $s(k)$ is the discrete sequence obtained after sampling $s(t)$.

There is a reverberation in the actual environment, where receiver 1 and receiver 2 receive signals, respectively

$$\begin{cases} x_1(k) = h_1(k) * s(k) + w_1(k) \\ x_2(k) = h_2(k) * s(k-\mathrm{D}) + w_2(k) \end{cases} \tag{2-3}$$

where $h_1(g)$ represents the room impulse response and $*$ represents the convolution operation. The analysis in this paper is a reverberation-based model that is close to the real environment.

3. ALGORITHM

3.1 Generalized Cross Correlation

Generalized Cross Correlation [4](GCC) on the basis of the CC algorithm, using $x_1(\mathrm{t})$ and $x_2(\mathrm{t})$ as the receiving signal, $x_1(\mathrm{t})$ and $x_2(\mathrm{t})$ are first processed by the pre-filter $H_1(f)$ and $H_2(f)$ respectively to obtain the output signals $y_1(\mathrm{t})$ and $y_2(\mathrm{t})$, and then as a general stationary signal using the Fourier function to transform the time domain signal $y_1(n)$ and $y_2(n)$ into the frequency domain processing, and the weighted function processes the cross-correlation function, making the main peak more sharp and prominent. Then use the Fourier inverse transformation to the time domain. This allows for more accurate latency estimation and stronger tracking of signals. The GCC algorithm flow diagram is shown in Figure 1.

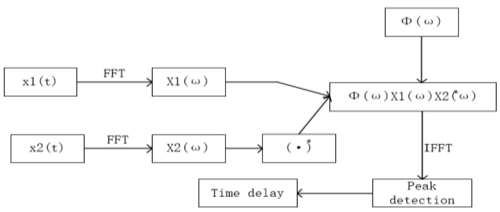

Figure 1 Schematic diagram of generalized cross-correlation algorithm flow

In the generalized cross-correlation weighting algorithm, in order to further improve the sharpness of the main peak and weaken other peaks, the GCC algorithm uses different weighted functions to process the cross-correlation power spectrum.

3.2 Homomorphic filtering

Homomorphic filtering[5], also known as reciprocal analysis, is a filtering technique used for anti-reverberation, where homomorphic filtering breaks down a signal into the smallest phase part and the full-pass component part. The minimum phase component of the signal has the characteristics of fast attenuation and small quadratic peak amplitude. The all-pass component of the signal provides position information useful for delay estimation. The reverberation intensity in the omni-pass component, especially early reverberation intensity, is greatly reduced. These features make the all-pass component of the signal not only retain the direct path delay information, but also reduce the reverberation effect. It can be seen that homomorphic filtering can improve the anti-reverb ability of the signal. The homomorphic filter decomposition method of the signal is shown in the following figure 2.

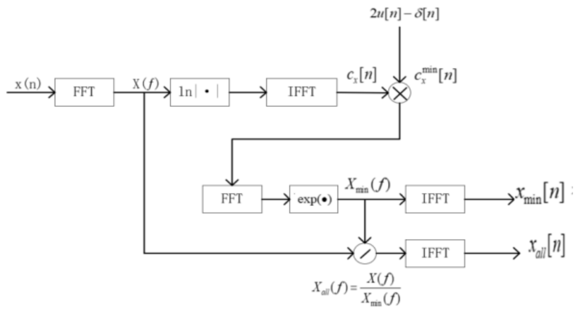

Figure 2 Decomposition of the minimum phase and full-pass components using homomorphic filtering.

The specific steps of the homomorphic filtering technology are shown in the figure above, and the received signal is homomorphic filtered by using the signal all-pass component to have the characteristics of anti-reverberation. After that, the all-pass components of the signal are cross-correlated to achieve the purpose of de-reverberation. However, since the minimum phase component is discarded during processing, the correlation between the signals is reduced, so consider using the secondary intercorrelation [6] to process the filtered signal to improve the correlation.

3.3 Hilbert transform

The Hilbert transform is to convert the correlation function of the even symmetry into odd symmetry, and the detection of the peak in the generalized mutual correlation is converted into the detection of the corresponding zero crossing point, and when the noise and reverberation interference cause the cross-correlation peak to become wider and flat, the delay can be estimated to a certain extent, and the noise immunity is certain. However, there is still a problem in judging the zero crossing point, and the influence of noise and other interferences makes it possible to fluctuate near the zero crossing point, resulting in multiple zero crossings, which delays the judgment on time, resulting in an increase in estimation error. In addition, when the signal sequence is too long, there are often multiple zero crossings, and in order to avoid this, other auxiliary algorithms are usually needed to detect it. However, this processing leads to a higher complexity of the peak detection algorithm.

Considering that the peak of the cross-correlation function corresponds to the zero crossing point of the Hilbert transform, a delay estimation algorithm for the Hilbert difference based on the intercorrelation is further proposed, that is, the difference between the cross-correlation function and the absolute value of the intercorrelation function after the Hilbert transformation[7]:

$$\mathrm{R}(\tau)=\mathrm{R}_{12}(\tau)-|\,hilbert(\mathrm{R}_{12}(\tau))\,| \qquad (3\text{-}1)$$

This algorithm can not only keep the value at the delay estimation of the cross-correlation function basically unchanged, but also suppress other pseudo-peaks of adjacent main peaks, solve the problem that adjacent pseudo-peaks and main peak amplitudes are similar, and also have better noise immunity. On the waveform, the peak of the intercorrelated waveform after the difference is sharper, which acts as a sharpening of the main peak and reduces the delay estimation error.

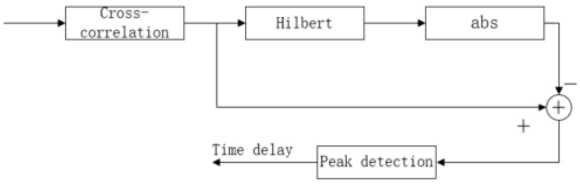

Figure 3 Schematic of Hilbert's delay estimation of the difference method.

The definition of the Hilbert transform is

$$\hat{R}(t) = \frac{1}{\pi}\int_{-\infty}^{\infty}\frac{R(t)}{t-\tau}d\tau \qquad (3\text{-}2)$$

3.4 New algorithm

This paper improves the algorithm block diagram as shown in the figure 4, the receiver receives the signal for inverted spectral domain filtering, to obtain the full-pass component of the signal, the full-pass component signal for secondary cross-correlation, before the cross-correlation function peak detection, the cross-correlation function for Hilbert difference processing, that is, the cross-correlation function and the Hilbert transformation and take the absolute value after the cross-correlation function subtraction, in order to achieve the effect of sharpening the main peak. Finally, the peak detection is performed to estimate the delay.

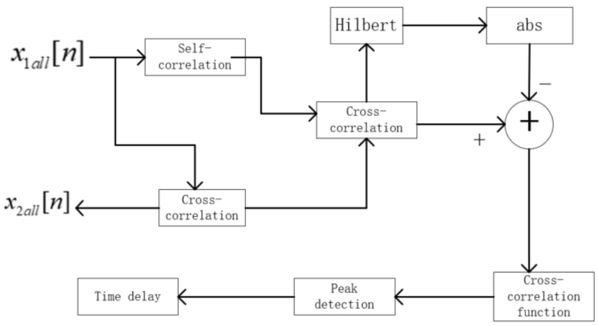

Figure 4 Block diagram of the new algorithm.

3.5 Simulation and analysis of results

In the simulation experiment, the length, width and height of the room are 10m, 5m and 3m, respectively, in the delay estimation algorithm, the delay difference between the two channels is estimated, and only the delay estimation of the signal received by the two microphones is required to verify the effectiveness of the algorithm.

During the simulation, a pure voice from the TIMT voice dataset is used, and the frequency range of the voice is 300Hz to 3400Hz, and the sampling frequency is 16kHz. The IMAGE method is used to simulate the pulse response of the room, simulating the reverberation caused by the absorption reflection of walls and objects in the room. In addition, there is the sound of tapping the keyboard, the hum of the air conditioner, and the boiling water of the water dispenser. These ambient noises are uncorrelated noises, which are simulated by adding Gaussian white noise with different signal-to-noise ratios to the signal.

The reverberation time in the indoor environment is usually 0.3 to 0.7s[], the following figure 5 is in the room small noise (signal-to-noise ratio 20dB), strong reverberation (the longer the reverberation time, the stronger the reverberation), the received signal mutual correlation obtained the result graph. where the abscissa is time and the ordinate is the normalized amplitude. The simulation diagram is, from top to bottom, the basic cross-correlation algorithm, the generalized cross-correlation algorithm and the cross-correlation function diagram of the new algorithm, and the main peak local diagram. By comparing function waveforms, you can compare the strengths and weaknesses of the new algorithm.

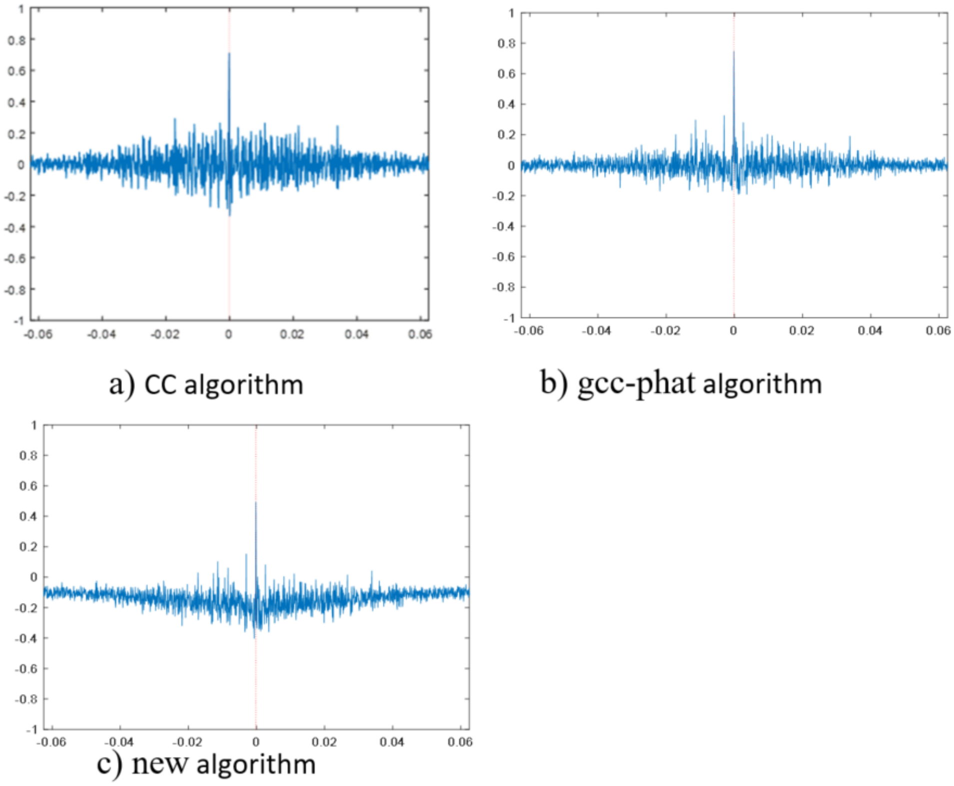

Figure 5 Algorithm comparison waveform plot in reverberation environment.

Comparing Figure 5, it can be seen that in the case of large reverberation intensity, there are more glitches near the peak of the basic cross-correlation function, and the peak is high, and the position of the main peak is error. The correlation algorithm after inverted spectral domain filtering reduces the number of spectral peaks near the main peak, and the correlation function waveform processed by the algorithm in this paper makes the main peak sharper, and the difference between the main peak and the nearby spectral peak increases, which is conducive to improving the delay estimation accuracy.

Similarly, Figure 6 is a simulation performed in an environment with high ambient noise interference (signal-to-noise ratio of -5dB) and small reverberation times. It can be seen that when the ambient noise is large, the main peak of the correlation function is very close to the noise, and the main peak is submerged in the noise. The new algorithm makes the peak of the main peak sharp and has a certain resistance to noise.

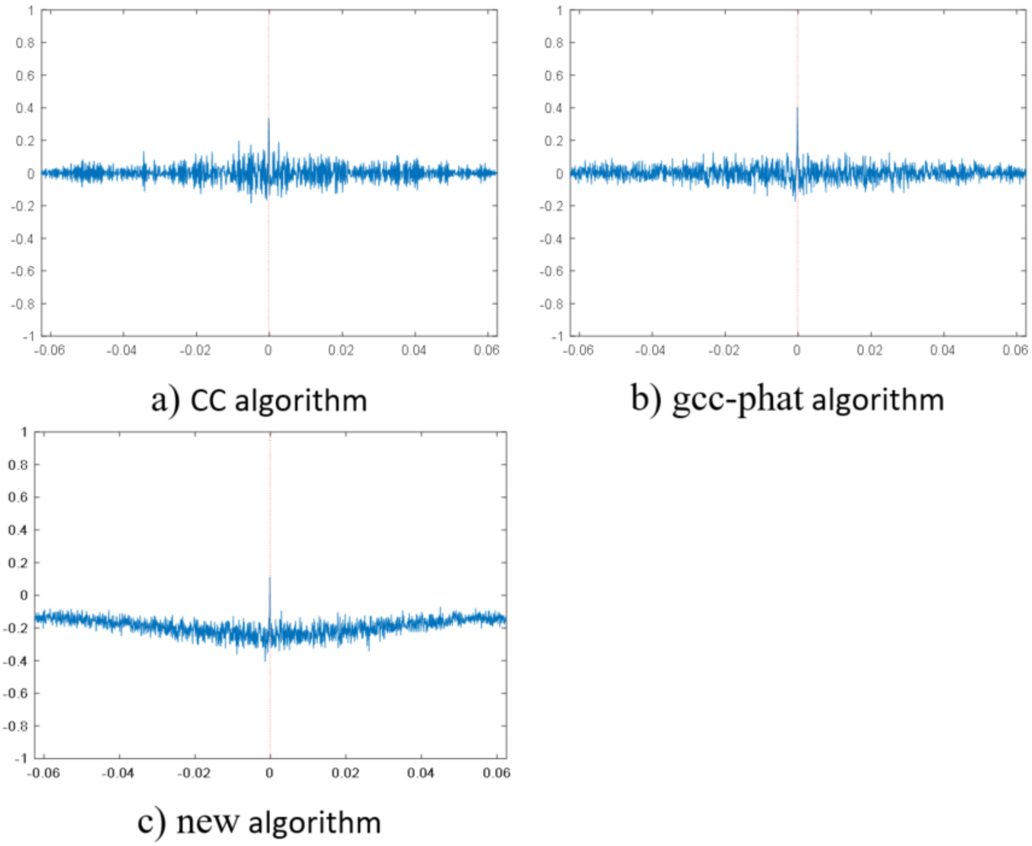

Figure 6 Algorithm comparison waveform plot in noisy environments.

In summary, the new algorithm has a good effect on anti-reverb and a slight improvement in anti-interference.

Figure 7 a is a graph of the mean squared error curve of delay estimation for each algorithm at different signal-to-noise ratios under low reverberation conditions (RT60=300ms).

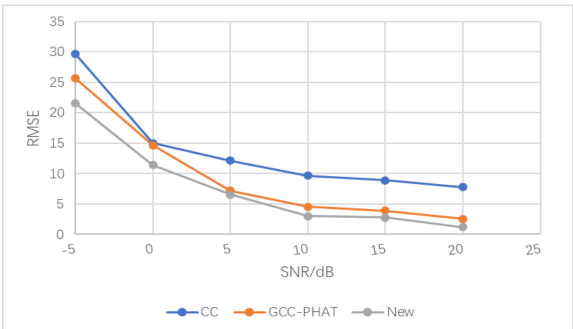

Figure7 The relationship between signal-to-noise ratio and delay estimation mean squared error.

According to Figure a, the CC algorithm and the GCC-PHAT algorithm are sensitive to noise, when the signal-to-noise ratio is less than 10dB, the delay estimation mean squared error of the CC algorithm is greater than 10%, in comparison, the noise immunity of the GCC-PHAT algorithm is slightly stronger than that of the CC algorithm. Compared with the traditional first two algorithms, the new algorithm has better noise resistance performance than high signal-to-noise ratio at low signal-to-noise ratio. However, on the whole, the new algorithm has a high signal-to-noise ratio, and the noise resistance performance is improved by 1% to 2%.

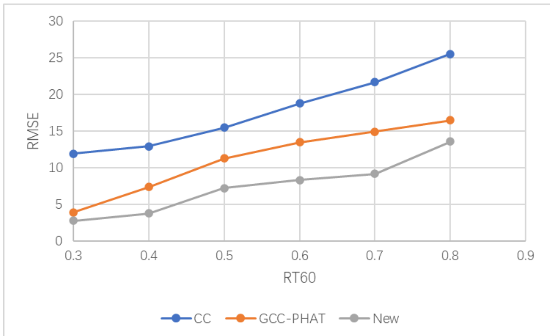

Figure8 Reverberation time and delay estimate the relationship between mean squared error.

4. CONLUSION

In this paper, after homomorphic filtering of the received signal, the correlation of the signal is weakened, which affects the accuracy of the delay estimation, and a new delay algorithm is proposed. Homomorphic filtering processing signal, the signal is divided into full pass component and minimum phase component, the signal all-pass component contains more direct sound part, and the minimum phase component contains more reverberation components, so the reverberation has almost no effect on the all-pass component, only the all-pass component part is mutually correlated to achieve the purpose of anti-reverb. The quadratic cross-correlation is used to improve the correlation degree of the signal, and the Hilbert difference is introduced to sharpen the cross-correlation peak and improve the delay estimation accuracy. Simulation results show that compared with the basic cross-correlation algorithm, the proposed algorithm reduces the rms error of delay estimation by 13% and the GCC-PHAT algorithm by 5%, which can weaken the influence of room reverb on delay estimation. In terms of noise immunity, the proposed algorithm is also improved compared with the traditional algorithm, which improves the noise immunity of the delay estimation algorithm.

5. ACKNOWLEDGMENTS

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