

A ROBUST METHOD FOR DELAY ESTIMATION IN MOBILE APPLICATIONS

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ABSTRACT

In these days voice and data communication is mostly through mobile devices. In wireless communications signals that arrive at base station via various paths. Path difference creates different delay, we have to estimate that delay, so that the mobile can be localized for emergency services. The proposed method is to find the Time Delay between transmitted and received signals using an Orthogonal Decomposer, which consists of all pass filter, moving average window and adaptive filter is used for this purpose. In this method we consider two windows one is the observation window and the other is detection window. In the observation window transmitted and received signal delayed by some amount are present. In the detection window the transmitted signal is detected and the delay between transmitted and received signals is estimated. In order to estimate the delay accurately the total delay is taken as sum of coarse delay and fractional delay. Coarse delay is based on integer which can be found by minimizing the coarse objective function. Fine delay is based on fraction which can be found by minimizing the fine objective function.

INTRODUCTION

Considerable amount of research went into time delay estimation for the past three decades. Previous techniques that estimate the difference in arrival time between two signals corrupted by noise, received at two separate sensors, are based on the determination of the peak of the generalized cross correlation between the signals. To achieve good resolution and stability in the estimates, the input sequences are first weighted. Invariably, the weights are dependent on input spectra which are generally unknown and hence have to be estimated

Problem of Delay estimation can be stated as calculating time delay between transmitted and received signal captured at receiver. A time delay may be defined as the time interval between the start of an event at one point in a system and its resulting action at another point in the system. During the last forty years, the problem of estimating the time delay between signals received at two spatially separated microphones in the presence of noise has been considered for a variety of applications such as in acoustics, radar communications, microphone array processing systems and speech recognition. This physical problem in two dimensions is

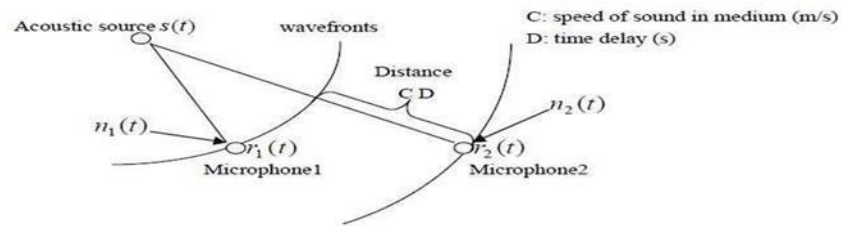


Fig. 1 Time-delay associated with two microphones

The received signal at the two microphones can be modelled by:

$$r_1(t) = s(t) + n_1(t)$$

$$r_2(t) = s(t - D) + n_2(t)$$

where $0 \leq t \leq T$, $r_1(t)$ and $r_2(t)$ are the outputs of two spatially separated microphones, $s(t)$ is the source signal, $n_1(t)$ and $n_2(t)$ represent the additive noises, T denotes the observation interval, and D yields the time delay between the two received signals. The signal and noises are assumed to be uncorrelated having zero-mean and Gaussian distribution. There are many algorithms to estimate the time delay. The Cross-Correlation (CC) method is one of the basic solutions of the TDE problem. Many other TDE methods develop based on this algorithm. The CC method cross-correlates the microphone outputs and considers the time argument that corresponds to the maximum peak in the output as the estimated time delay. To improve the peak D detection and time delay estimation, various filters or weighting functions have been suggested to be used after the cross correlation. The estimated delay is obtained by finding the time-lag that maximizes the cross-correlation between the filtered versions of the two received signals. This technique is called Generalized Cross-Correlation (GCC). The GCC method is the most popular technique for TDE due to their accuracy and moderate computational complexity. The role of the filter or weighting function in GCC method is to ensure a large sharp peak in the obtained cross-correlation thus ensuring a high time delay resolution. There are many techniques used to select the weighting function; such as the Roth Processor, the Smoothed Coherence Transform (SCOT), the Phase Transform (PHAT), the Eckart Filter, and the Maximum Likelihood (ML) estimator. They are based on maximizing some performance criteria. These correlation-based methods yield ambiguous results when the noises at the two sensors are correlated with the desired signals. To overcome this problem, higher-order statistics methods were employed. There are also some other algorithms used to estimate the time-delay. The matching x and s (MXS) and matching s and x (MSX) methods compare the cross-correlation of the received signals $r_1(t)$ and $r_2(t)$ with the auto correlation of the reference signal $r_1(t)$. Algorithms based on minimum error: Average Square Difference Function (ASDF) and Average Magnitude Difference Function (AMDF), seek position of the minimum difference between signals $r_1(t)$ and $r_2(t)$. Adaptive algorithms such as LMS can also be introduced into the TDE. In these algorithms, the delay estimation process is reduced to a filter delay that gives minimal error. Nowadays, many other methods are employed in the TDE, such as, MUSIC, ESPRIT, and Wavelets.

PROPOSED METHOD:

Delay Estimation using Orthogonal Decomposer

A lot of research has gone into time delay estimation for the past three decades [1],[2],[3],[4],[5]. Previous techniques that estimate the difference in arrival time between two signals corrupted by noise, received at two separate sensors, are based on the determination of the peak of the generalized cross correlation (GCC) between the signals. To achieve good resolution and stability in the estimates, the input sequences are first weighted. Invariably, the weights are dependent on input spectra which are generally unknown and hence have to be estimated. The proposed method accurately estimates the time delay even under poor SNR conditions and does not require the knowledge of input spectra.

Data Generation

In order to estimate the delay between transmitted and received signals the transmitted narrow band signal s_k can be generated using

$$s_k = (m_k * u_k) + \gamma_k$$

where u_k is a sinusoidal signal defined as $\cos(2\pi f_n k)$ and γ_k is an additive zero mean white function representing the physical process. The operator $*$ represents convolution. Received signal is delayed and uses another modulating function making it different from transmitted signal. Narrow Band (NB) Tx or Rx signal $s(t)$ is processed as where δt is the sampling time and k is an integer.

$$s^{Tx}(t) = s^{Tx}(k\delta t) = s^{Tx}\{k\}$$

are observed over a period of time in the handset. Here we consider two windows one is the observation window and the other is detection window. In this observation window both transmitted (Tx) and received (Rx) signal delayed by $\Delta\tau$ are present. We estimate the detection window from this observation window, which is a subset of the observation window.

Detection Window

Orthogonal Decomposer¹ [6] (see Fig 3.1) consists of all pass filter, moving average filter and adaptive filter. The narrow band signals is passed through an orthogonal decomposer.

This process is shown by equations below. In this process the radial pole position r_k in equation is constantly adjusted to meet the demand of creating phase shift of π between s_k and v_k .

$$v_k = r_k v_{k-1} - r_k s_k + s_{k-1}$$

$$z_k = s_k v_k$$

$$r_k = r_{k-1} + e_k$$

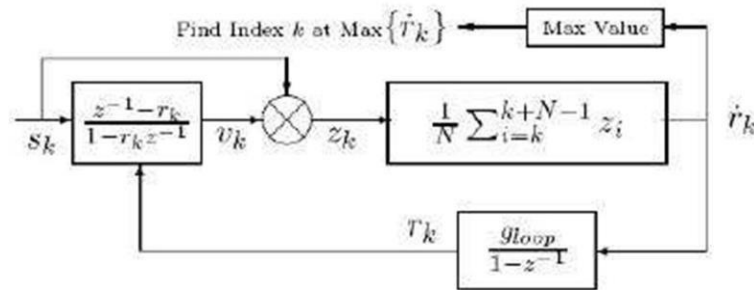


Figure 2: Adaptive All Pass Filter

$$u_k = \cos(2\pi f_n k + \phi)$$

$$x_k = r_k x_{k-1} - r_k u_k + u_{k-1}$$

$$z_k = u_k x_k$$

$$\dot{r}_k = r_{k-1} + e_k$$

where gloop is the loop gain of the adaptive filter, r_k is the pole position of the all-pass filter (APF) and N is the length of moving average (MA) filter to obtain the DC component. Fig 3 depicts the adaptive all pass filter given by the above equations. The threshold between short-term and long-term depends on the application, and the parameters of the moving average will be set accordingly. For example, it is often used in technical analysis of financial data, like stock prices, returns or trading volumes. It is also used in economics to examine gross domestic product, employment or other macroeconomic time series. Mathematically, a moving average is a type of convolution and so it can be viewed as an example of a low-pass filter used in signal processing. When used with non-time series data, a moving average filters higher frequency component without any specific connection to time, although typically some kind of ordering is implied. We obtain accurate value of r that minimizes the function JF . Substitute the value of r in $\delta\phi$ to obtain its value. Finally substitute the estimated parameters N and $\delta\phi$ to obtain the delay $\Delta\phi$. The algorithm for estimation of Time Delay is tested using MATLAB. Data is generated and processed by means of MATLAB. The processing code that estimates the Time Delay is written and compiled in MATLAB software using mex compiler to get DLL files. DLL files or dynamic link library files ensure security to code files and faster execution. Results obtained are discussed below.

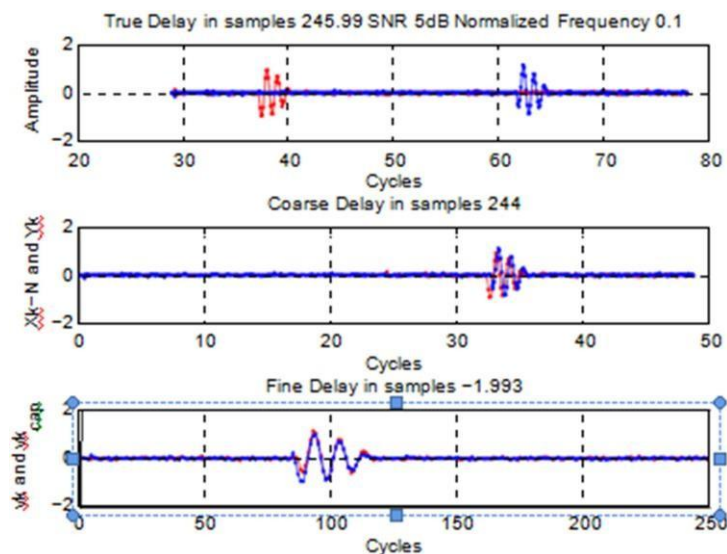


Fig. 3 Detection Window, Coarse Delay and Fine Delay Computation

The actual time delay is 245.990, Estimated time delay is 245.9930, Percentage of error is - 0.0012. The performance of the algorithm is tested for variable sampling rate and variable delay.

Variable Sampling Rate:

Performance of the algorithm for various $4 < 1/f_n < 9.6$ for a fixed delay of 4.8 Cycles is computed and is depicted in below fig. It may be noticed that accuracy is ± 2 if the Samples/Cycle is between 5 and 7 while the accuracy is excellent around 6 Samples/Cycle

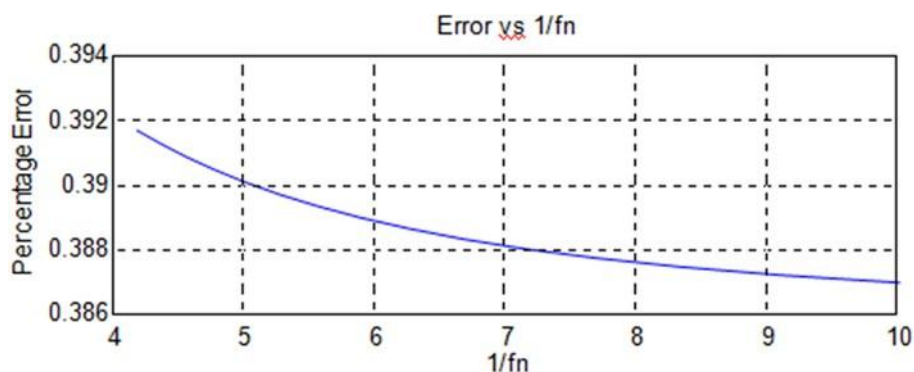


Fig. 4: Percentage Error variation with Sampling Rate

Variable Delay

Sampling rate is fixed at 6 samples/Cycle and delay is varied from 0.8 Cycles to 8.4 Cycles and the error in estimate is presented. The error is contained below 1.5, showing considerable promise.

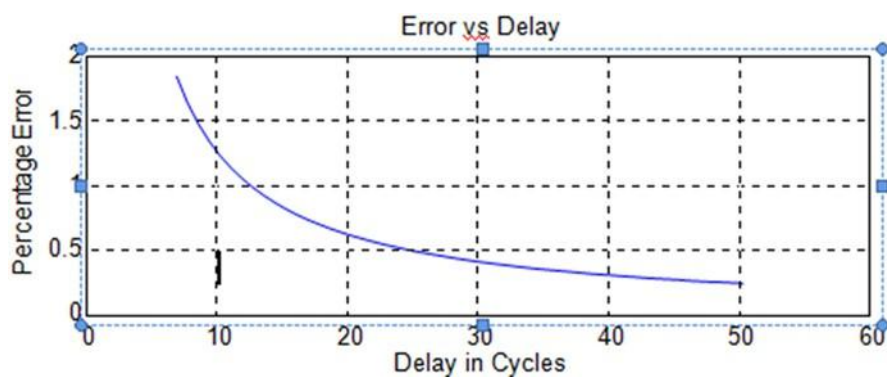


Fig. 5 Percentage error variation with delay

Conclusion

This paper presents a novel method for estimating the delay. An algorithm is designed for implementing this method in an embedded system using Orthogonal Decomposer. Initially, a digitized narrow band signal is transmitted and received, the received signal is the delayed version of transmitted signal. In this method, two windows are taken: one is the observation window and the other is the detection window. In the observation window, both transmitted and received signals delayed by the time delay $\Delta\tau$ are present. The detection window can be estimated from the observation window by finding detection window indices using orthogonal decomposer. The delay $\Delta\tau$ is taken as some of integer delay and fractional delay, and these two delays are found out to obtain the total delay. Computational burden and this seamless integrated algorithm with detection and estimation is the main focus. Error is less than 1.5 for a given sampling rate.

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