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Novel Acoustic Time Delay Estimation Model and Experimental Verification

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ABSTRACT

The bubble formed due to absorption of neutron energy in superheated emulsion based neutron detector generates the acoustic signal. There is always possibility of false bubble counting due to external acoustic noise registration. The developed model estimates acoustic signal time difference of arrival (TDOA) at two microphones/transducers. The time delay estimation model is developed and its experimental verification is carried out. The model depends on the four main parameters i.e. azimuth angle, elevation angle, spatial separation between the sensors (micro-phones) and distance between the sensor and acoustic source. Dependency of the model on ratio of distance between microphone & acoustic source (r) and spatial separation between microphones (d has been also emphasized. The model is helpful in development of electronics for the above detector to avoid the interference of external acoustic noise registration.

Key word: Time delay estimation, acoustic source, superheated emulsion detector.

INTRODUCTION

Time delay estimation (TDE) between signals received at two microphones has been proven to be a useful parameter for many applications. Speech enhancement, speaker localization, speech and speaker recognition and meeting activity detection are some examples of applications based on TDE. Five different time delay estimation methods are important. These methods are cross-correlation (CC), phase transform (PHAT), maximum likelihood estimator (ML), adaptive least mean square filter (LMS) and average square difference function (ASDF).

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 communication, microphone array processing systems and speech recognition. This physical problem in two dimensions is shown in Figure 1.

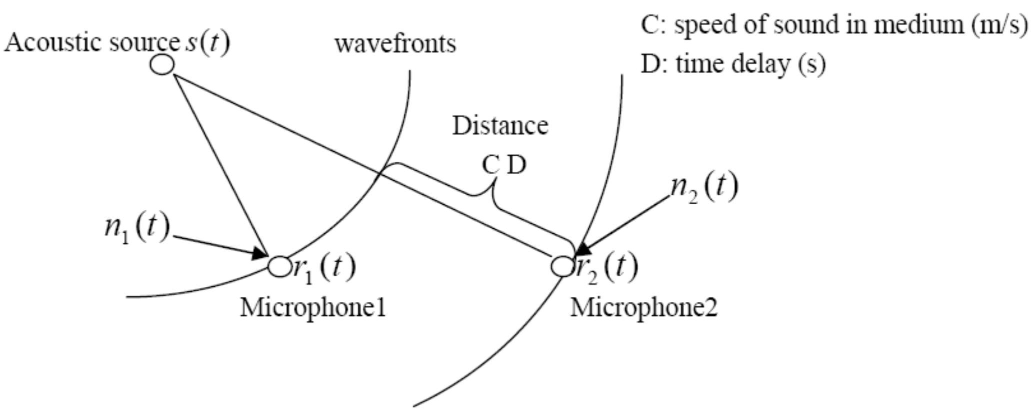

Fig (1) Time-delay associated with two microphones

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

$$\begin{aligned} r_1(t) &= s_1(t) + n_1(t), \\ & \qquad 0 < t \leq T \\ r_2(t) &= s_2(t - D) + n_2(t) \end{aligned} \qquad (1)$$

Where 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 D [1]. The cross-correlation (CC) method is one of the basic solutions of the TDE problem [1]. 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 detection and time delay estimation, various filters, or weighting functions, have been suggested to be used after the cross correlation [2]. 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) [2]. The GCC method, proposed by Knapp and Carter in 1976, 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 [1, 2]. 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 [1, 3]. 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 [4] compare the cross-correlation of the received signals r_1 (t) and r_2 (t) with the autocorrelation 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) [4]. Adaptive algorithms such as LMS can also be introduced into the TDE [6]. 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 [7] and wavelets [8]. However, time-delay estimation is not an easy task because it may face some practical problems, such as, room reverberation, acoustic background noise and the short observation interval.

Defence laboratory, Jodhpur is working on development of superheated emulsion drop detector since last 15 years [9-10]. The Superheated Drop Detector (SDD) [11] is a homogenous suspension of super heated Freon droplets inside a viscous elastic gel, which may undergo transitions to the gas phase upon energy deposition by incident radiation. Each droplet behaves as a micrometric bubble chamber. SDD have been widely used in neutron dosimetry [12-13] and spectrometry [14-16]. They have shown to comply with ICRP 60 recommendation of measurement, real time response, low minimum detection threshold and most importantly, a nearly dose equivalent response. Superheated emulsion detector based bubble reader system is developed indigenously. There is problem in instrumentation of the detector is that acoustic noise registration taking place with the actual bubble signal. This acoustic noise makes possibility of incorrect neutron dose and neutron dose rate estimation. The reader system is very useful in case of nuclear emergency disaster management. Implementation of this time delay model in electronic design may improve the performance. This theory will be helpful in estimating the maximum time delay. Two Piezo electric transducers/microphones are placed in the reader system. One transducer attached with the neutron sensor for counting of bubble formation due to neutron exposure. Second one is placed for monitoring for the environment acoustic noise in form of sound or vibration. If there is environmental nose present then it will affect the both transducers. The output of second transducer is applied to an electronic switch through which the operation of the counter will be stopped. Thus registration of external acoustic noise can be controlled. To resolve this problem time delay estimation is compulsory need. This problem has been resolved by this model. Designs of the electronic circuits by considering this model are under progress.

MATERIALS AND METHODS

(A) 2-D model

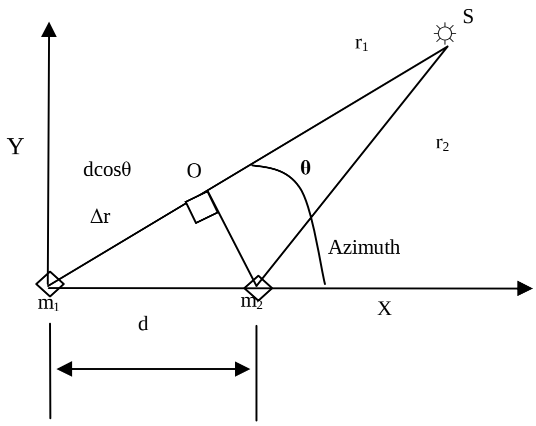

Fig (2) 2-D model

Two microphones are mounted on same surface (base plate) and having separation of d. Microphone1 (m_1) attached with neutron sensor for counting of the bubbles formed due to neutron dose. Microphone2 (m_2) is attached for same base plate for sensing of environmental acoustic noise. r_1 is the path length acoustic signal from acoustic source to the microphone phone 1 and r_2 is the path length for acoustic signal from acoustic source to microphone 2. Δr was the difference between both path lengths.

$\Delta r = r_1 - r_2$, Velocity of sound wave (c) =distance / time,

$c = \Delta r / \Delta t$, $\Delta t.c = \Delta r$,

$\cos\theta = \Delta r/d$ ($m_2O \perp m_1S$),{ Where straight line m_2O is perpendicular to m_1S}

$\Delta r = d\cos\theta = \Delta t.c$, {Where θ (azimuth angle) is the angle between reference line(X axis) and path line r_2 of acoustic signal }

$\Delta t = (d/c)\cos\theta$, $\Delta t = (d/330).\cos\theta$ {where d is spatial separation between two micro-phones) {Where c = 330 m/s (air)}

$\Delta t = 3d\cos\theta$ (ms), $D = 3d\cos\theta$ (ms) (2)

(B) 3-D model

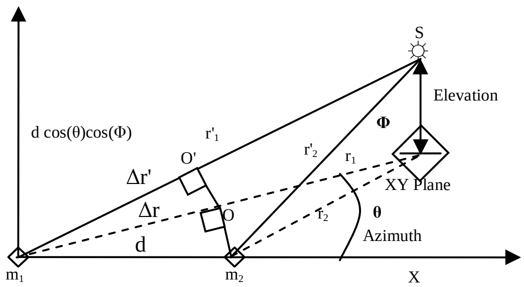

Fig (3) 3-D Delay model

For 3-D azimuth angle (θ) is and elevation angle Φ (angle between the line r_1 and r') taking variables.

$\Delta r = r_1 - r_2$, velocity (c) = distance/time
$c = \Delta r/\Delta t$, $\Delta t.c = \Delta r$
$\cos\theta = \Delta r/d$ ($Om_2 \perp m_2S$)
$\cos\Phi = \Delta r'/\Delta r$ ($OO' \perp m_1S$)
$\cos\Phi = \Delta r'/d\cos\theta$ (from eq. 2)
{Where Φ=Elevation angle}
$\Delta r' = d\cos\theta\cos\Phi = \Delta t.c$,
$\Delta t = (d/c)\cos\theta\cos\Phi$
$\Delta t = (d/330).\cos\theta\cos\Phi$
$\Delta t = 3d\cos\theta\cos\Phi$ (ms)
$D = 3d\cos\theta\cos\Phi$(ms) (3)

Case I

If both θ and Φ angles are set at 90° degree. Then the time delay will be zero (cos90°=0) as per equation 3.this is the condition of minimum delay. It is concluded that both the microphones received that signal at a time. Arrangement scheme of microphones and acoustic source for minimum delay is shown in the fig (4).

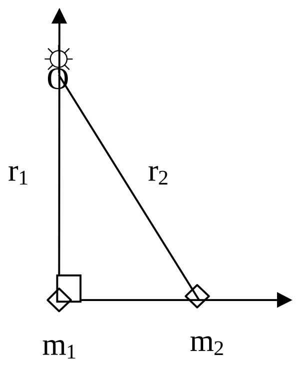

Fig (4) Arrangement of microphones and acoustic source for zero delay

Case II

If both θ and Φ angles are set at 0°. Then the time delay will be directly proportional to d (cos0°=1) as per the equation 3. This is the condition for maximum delay. Arrangement scheme of microphones and acoustic source for maximum delay is shown in the fig (5).

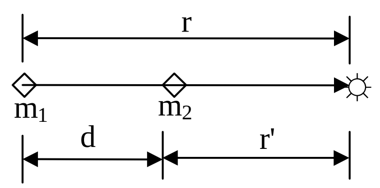

Fig (5) Arrangement of microphones and acoustic source for maximum delay for fix d

Experimental set up

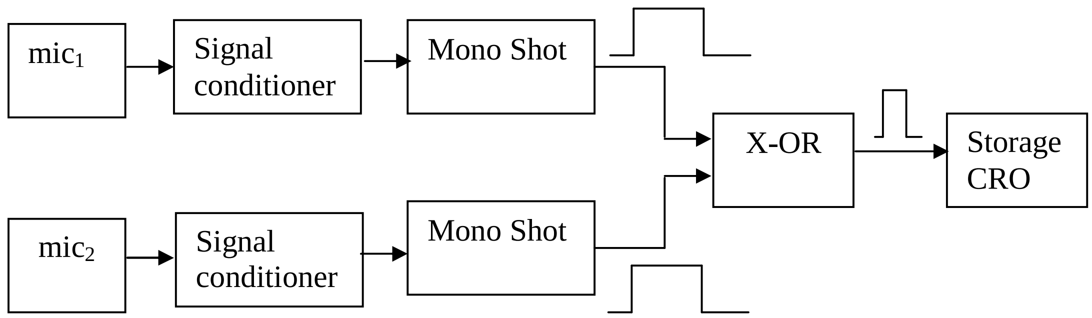

Fig (6) Electronics circuit for measurement of time delay in arrival of signals

Fig (6) shows the experimental set up for measurement of time difference in arrival of acoustic signal at microphones. Two microphones/transducers were placed on same plane having spatial separation d. Signal conditioning circuit is developed to process the output signal of the transducer. The circuit consists of pre-amplifier, band pass filter and comparator. The signal conditioner circuit reduces the noise level & improves the signal strength. The output of the signal conditioner is applied to the mono-shot to make proper output in form pulse. The biasing and processing units for both the transducers are similar in nature. The out put of the mono-shots are applied to the X-OR gate. The output of the X-OR gate is a pulse. The time period of the pulse is the measure of time difference between the signals. If output of X-OR gate is zero it indicates that both microphones received the acoustic signal at same time. The output of the X-OR gate is applied to the storage oscilloscope for measurement.

Experiments were carried out for verification of the theory. Two microphones/transducers and one acoustic source were used to perform the experimentation. Both the transducers are mounted on the same base plate such as spatial separation between the transducer was adjustable. Buzzer was used as acoustic source which biased by sharp electric pulse. Microphone mic_1 was placed at the center of circle with radius of 1m and the source (buzzer) was placed on the periphery of circle from 0° to 360° at steps of 30°. The experiment carried out from 0° to 180° because from 180° to 360° cosine has the same values but has opposite trend. During all the experiment the distance between acoustic source and sensor was kept 1meter.

For 3-D model same experiments were carried out by fixing the azimuth angle at 30° and varying the elevation angle from 0° to 180° in step of 30°. In continuation of this experimentation was also carried out by fixing the θ (azimuth angle) and Φ (elevation) at 0° and source at distance of 1 meter from transducer mic_1. In this case spatial separation between the transducer is assigned as a variable.

All the experimentations were carried out by following assumptions

1. Single sound source, infinitesimally small, Omni directional source.
2. Reflections from bottom of the plane and from the surrounding objects are negligible.
3. No disturbing noise source contributing to the sound field.
4. The noise source to be located, is assume to be stationary during the data acquisition period.
5. Microphones are assumed to be both phase and amplitude matched and without self noise.
6. The change in sound velocity due to change in pressure and temperature neglected. The velocity of sound in air is taken as 330 m/sec
7. Knowledge of positions of acoustic receivers and perfect alignment of receivers as prescribed by processing techniques.

RESULT AND DISCUSSION

Theoretical values have been calculated by considering equation 1 & substituting the value theta from 0° to 180° and value of spatial separation (d=5 cm) for 2 D model. In 3-D model substituting the value of Φ from 0° to 180° & θ fixed at 30 ° and spatial separation d at 5 cm in equation 3.

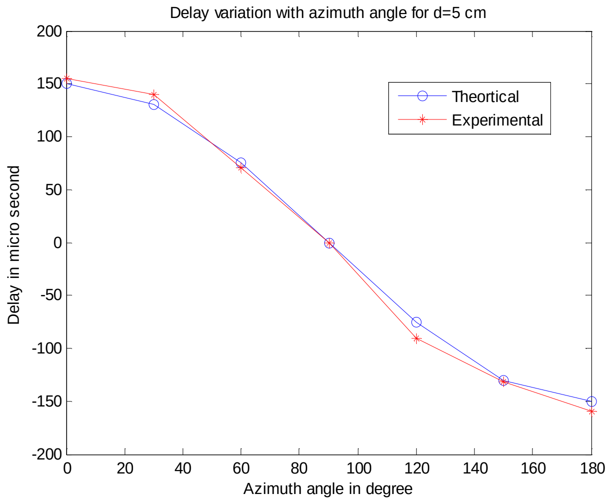

Fig (7) Delay variation with azimuth angle

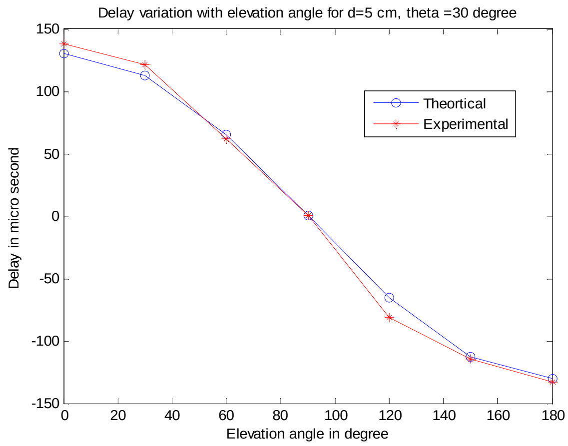

Fig (8) Delay variation with Elevation angle

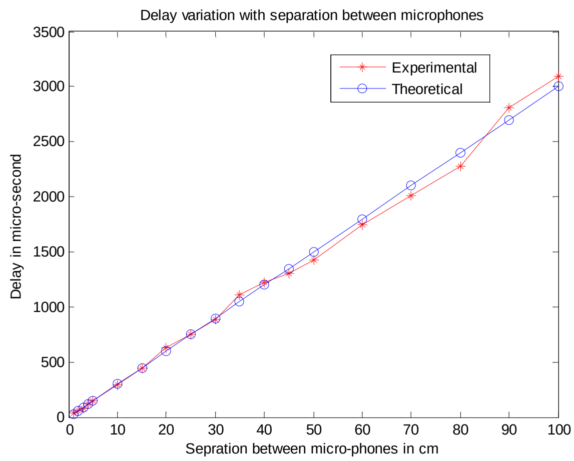

Fig (9) Delay variation with separation (spatial) between transducers

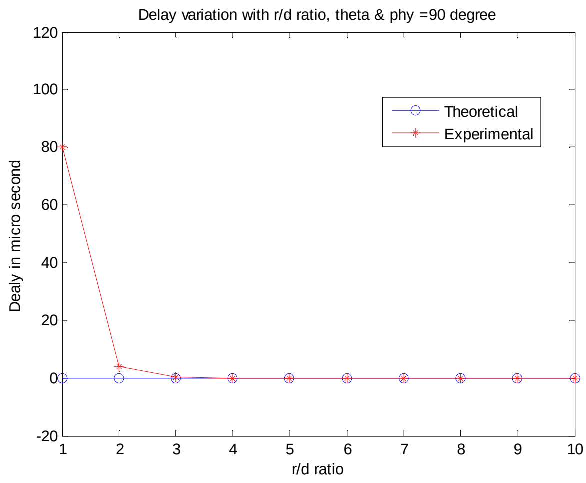

Fig (10) Delay variation with r/d ratio for minimum delay case

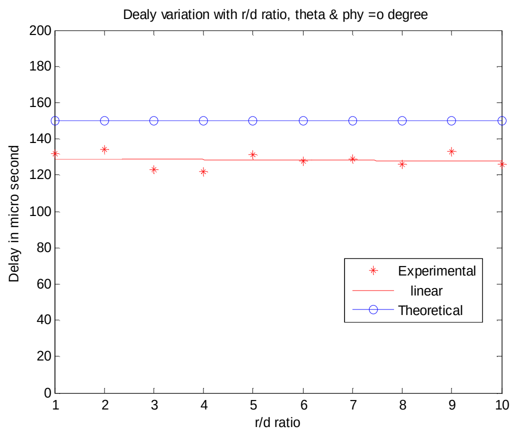

Fig (11) Delay variation with r/d ratio maximum delay case

The fig (7) & fig (8) shows the variation in the delay with azimuth angle and elevation angle keeping the separation constant. The curve shows that there is linear relationship between the theoretical and experimental data. This linear and comparable nature of theoretical and experiment curve validate the model. Variation of the delay with separation between transducers is shown in the Fig (9). Fig (10) & Fig (11) show the relation between delay and r/d ratio for minimum and maximum delay condition.

CONCLUSION

The theoretical and the experimental results of 2-D and 3-D delay models are comparable. The trends of both curves are same proves its validity. The models validation depends on the r/d ratio. The models are validated when path length distance of acoustic signal should be at least 3 three time of the spatial separation between the transducers.

Perfect solutions are not possible since the accuracy depends on following factors viz. geometry of microphone and source, accuracy of microphone setup, Uncertainty of microphone setup, lack of synchronization of the microphones, inexact propagation delays, presence of noise source and numerical round off errors.

The delay model will be utilized to estimate the maximum time difference in arrival of acoustic noise signal from unknown location for superheated emulsion detector based reader system. Electronics for reader system will be redesigned by considering the model such that in acoustic noise condition the instrument may stop the bubble counting to avoid noise registration possibility. Redesign of the electronics hardware for the reader system is under progress.

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