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Study of Microphone Array Characteristics and Noise Reduction

Dr. Navin Kumar [1], Dr. Alka Singh[2]

[1]Centre Coordinator, IGNOU Centre for Engineering, Muzaffarpur, Bihar, India.

[2] Assistant Professor, Department of Pure and Applied Physics,
Guru Ghasi Das Central University, Bilaspur, Chhatisgarh, India.

Abstract

The use of microphone array for speech signal capturing provides many benefits over the use of single microphone for example it captures spatial features of the speech signal, it improves SNR of the signal and it can be used to steer response in different directions etc. Microphone arrays are used in many state of the art acoustic signal processing technologies such as in beamforming, ASR, speech signal separation etc. The response of the array depends on geometry of the microphone array too. In different applications microphone array of different geometrical shapes are used. The most common geometrical arrangements of microphones are linear, circular, triangular and spherical form of arrays. For different geometry of the array, the geometrical constraints for signal processing changes. In the present paper different microphone array characteristics have been studied and noise reduction and improvement in SNR by using microphone array have been established.

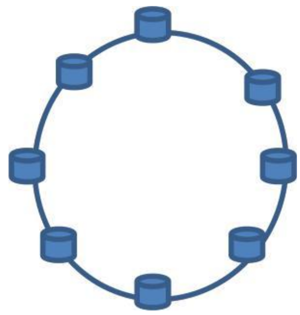
Keywords: Microphone array, Acoustic signal processing, DOA estimation

INTRODUCTION

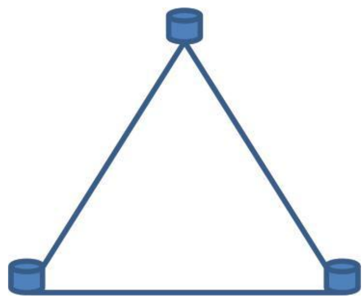
Human beings are gifted with two ears to listen sound and are capable in extracting many information such as about speaker and his/her locations, distance and direction of active speaker in addition to language dependent information from sound of a speaker from the sound waves reaching at ears. It is well researched that the sound signals reaching at two ears contain temporal and level differences known respectively as Inter-aural Time Difference (ITD) and Inter-aural Level Difference (ILD). These two cues are used by our hearing system to find out DOA of the acoustic source [1] [2] . The two ears act as sensor to pick up speech signal from two different points of observations in the acoustic field. Imitating the same there have been uses of multiple sensors (microphones) to pick up the sound with different spatio-temporal characteristics. Such multiple microphone arrangements in fixed geometric framework for speech signal pick up is named as microphone array [1] [5] .

Obviously, it is not simply spatial distribution of the microphones rather some fixed geometrical patterns are used to place microphones, keeping positions of the microphone fixed during signal pick up. However, in the active audition the position of sensor may change during the speech signal pick up like that of human being. The idea of use of microphone array in the speech signal processing is not inborn rather borrowed from the area of radar signal processing [3] where the general framework of array signal processing developed much before. The shape of microphone array depends on used geometric pattern and accordingly, there have been developments of different microphone arrays such as linear microphone array, circular microphone array, triangular microphone array, spherical array, etc [1] [6] . Here, we will use the linear microphone array in which sensor elements (microphones) are linearly distributed with equal inter-element distances.

(a) Uniform Linear microphone Array (ULA)



(b) Circular Microphone Array



(c) Triangular Microphone Array

Figure 1.1 Microphone array with different geometry. (a) ULA in which all the elements are equidistant and collinear (b) CMA in which elements are placed along circumference of a circle (c) TMA elements or microphones are placed at vertices of a triangle.

Thus microphone array captures speech signal with different spatiotemporal characteristics. The Microphone array represents spatial filter sources. The signal captured by microphone array is processed using different techniques either in time domain or in frequency domain to estimate different information such as DOA, speech from a particular speaker (similar to steering hearing attention), distances of the speakers etc.

Microphone array's behavior is different from a single microphone rather combination of multiple microphones.

It has modification in directional gain due to which microphone arrays offer better Signal-to-Noise Ratio (SNR) and Signal-to-Reverberation Ratio (SRR) than a single microphone can Ambient Noise Gain: The isotropic ambient noise gain for a given frequency is the volume of the microphone array beam and can be given by

$$G_{AN(f)} \frac{1}{v} {}_{V} B(f,c)dc \text{ Where: } \quad (1.1)$$

V = microphone array work volume *B* (*f, c)* = directivity pattern of array.

a. Ambient noise gain represents the proportion of the noise floor RMS in relation to the output of the microphone array and to the output of an omnidirectional microphone. A lower value of it is better, and 0 dB means that the microphone array does not suppress ambient noise at all.

b. Weighted Ambient Noise Gain: A-weighted ambient noise gain gives the proportion of the noise floor in relation to the output of the microphone array and to the output of an omnidirectional microphone as they would be compared by a human. For example, -6 dB NGA means that a human would say that the noise on the output of a microphone array is halve of an omnidirectional microphone.

c. Directivity Index; The directivity index characterizes how well the microphone array detects sound in the direction of the MRA while suppressing sounds that come from other directions, such as additional sound sources and reverberation. The *DI* is measured in decibels, where 0 dB means no directivity at all. The DI can be given by

$$DI = 10log_{10} \int_{f=10}^{Fs/2} \frac{P_T f}{Pf} df, \quad (1.2)$$

Where

$\overline{P}\ f = \int_{-\pi}^{\pi} \int_{-\pi/2}^{\pi/2} P(f,\phi,\theta)\cos\theta d\theta d\phi$ = average power over all directions,

$P_T\ f = P(f,\phi_T,\theta_T)$ = Power in the Main Response Axis

d. Noise Robustness Factor (NRF): This term indicate effect of noise on the resolution of DOA estimate by an array. For the two element linear microphone array the signal captured by each microphone in frequency f is delayed which is given by

$$\tau = \frac{d\sin\theta}{c} \quad (1.3)$$

The phase difference contain difference contain frequency and time delay which is function of q. Thus for the given frequency f and spacing d between the elements, DOA q. can be estimated by estimating above time delay from signal. In the presence of noise the estimated delay derivates randomly and accordingly the estimate value of DOA q. In order to measure sensitivity q estimation w.r.t. noise NRE can be defined as

$$NRE = I(\theta) = \frac{1}{\frac{d\theta}{d\tau}} \quad (1.4)$$

for the linear array NRE is given as

$$I_{ULA}(q) = \cos q, \quad (1.5)$$

C. Directivity-Pattern or beam pattern: This term fully characterize input output behavior of the microphone array. The amount of signal received by an array depends on location of source and orientation of array. Such property of array can be expressed in terms of aperture function. The signal received by a microphone from source at *9* is given by

x(n) = s(n)* h(n) (1.6)

Where h(n) impulse response between source and sensor

s(n) = signal source.

In the frequency domain it is given as

X(*f*) = A(*f,* q)S(*f,* q) (1.7)

The function A (*f,* q) is known as aperture function for the microphone array. For the n element linear microphone array it is given by.

$$A(f,\theta) = \left[1\ e^{-12\pi f2dc^{-1}\sin\theta} \ldots e^{-12\pi f(n-1)dc^{-1}\sin\theta}\right] \quad (1.8)$$

where d = inter-element spacing , c = velocity of sound.

Thus the aperture function defines response of the array as function of spatial, position. The amount of signal seen by an array depends on location of sound source w. r. t. microphone array. If the aperture function is plotted as the function of DOA and frequency it is known as the directionally pattern or beam pattern of the array. For the far field situated the directionally pattern of receiving aperture (Microphone array) is given by

$$D(f,k) = \int_{\infty}^{\infty} A(f,r)e^{i2\pi kr}dr \quad (1.9)$$

Where r= x y z^T = location of source along aperture

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k = direction vector of wave = $\frac{1}{\lambda}\sin\theta.\cos\phi\cos\theta$ (1.10)

The angle θ and ϕ are as shown in Figure 1.5.

Now if a linear array of length L is placed across x-axis with mid-point origin the DP is given by < <

Above expression is valid for far field situation and can be evaluated for any value of k_r However, it is practically bounded by 1 k_r 1. This interval is known as visible region of the operation. For a linear array if we consider a uniform aperture function,

$$D(f,x_a)L\frac{\sin(x_a L)}{x_a L} \qquad (1.11)$$

the normalized directivity pattern of the linear array is given by

$$D(f,\theta,\phi)=\frac{c\sin(Lf\sin\theta\cos\phi)}{Lf\sin\cos\phi} \qquad (1.12)$$

For the far field situation |r | > $2L/\lambda$. For the broade side source $\theta=\frac{\theta}{2}$ and for the end fire source ϕ = 0, or ϕ = p .

The DP of array is plotted in Array manifold: In the array signal model, the array steering vector represents complex array response to unit nplitude plane wave impinging from direction

It is response of the array for front side positions. The complex vectors of array steering matrix are also nown as manifold vector associated with a articular source situated in a particular direction *"t.* Obviously, the manifold vector for a particular direction eeps geometrical information that can describe fall of a plane wave front on the array from the same irection. The concept of array manifold is very essential to understand functioning of many array processing lgorithms. For the setup of N-sensors and M-distinct sources, the array response and is NxM matrix in which each column is an N- dimensional manifold vector. The well known algorithm MUSIC for DOA estimation looks for manifold vector which are orthonormal to the estimated noise-space. The properties of array manifold can be used to explain the behavior of Microphone array. Some important properties of array manifold one listed below, which is plotted in Figure 1.4.

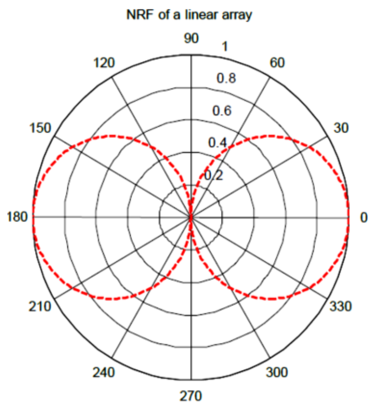

Figure 1.4.

The angle θ and ϕ are as shown in Figure 1.5 .

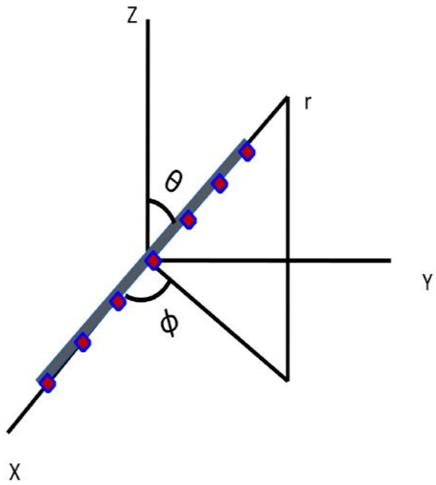

Figure 1.5 ULA along X-axis and sound source at r.

Array manifold: In the array signal model, the array steering vector represents complex array response to unit amplitude plane wave impinging from direction it is response of the array for front side positions. The complex vectors of array steering matrix are also known as manifold vector associated with a particular source situated in a particular direction θ_i. Obviously, the manifold vector for a particular direction keeps geometrical information that can describe fall of a plane wave front on the array from the same direction. The concept of array manifold is very essential to understand functioning of many array processing algorithms. For the setup of N-sensors and M-distinct sources, the array response and is NxM matrix in which each column is an N- dimensional manifold vector. The well known algorithm MUSIC for DOA estimation looks for manifold vector which are orthonormal to the estimated noise-space. The properties of array manifold can be used to explain the behavior of Microphone array. Some important properties of array manifold one listed below.

Array-Bandwidth: The array bandwidth is an important parameter of the array useful in devising capturing and processing of broadband signal. This is aparameter of array represents bandwidth of array transfer function. For an ULA of N microphones the array transfer function is given by

$$H(\omega\tau)=\frac{\sin(N\omega\tau/2)}{N\sin(\omega\tau/2)} \qquad (1.13)$$

and the bandwidth is given by

$$BW=\frac{2\pi c}{Nd\sin\theta} \qquad (1.14)$$

Obviously, the bandwidth varies as function of DOA for the ULA.

For the source at broadside on position, the bandwidth of the array becomes infinite and when the source is at end-on position it is $2\pi c/Nd$.

RESULTS AND CONCLUSION

A higher number means better directivity. For example an ideal cardioid microphone should have *DI* of 4.8 dB, but in practice cardioid microphones have a *DI* below 4.5 dB.

i. Manifold curve For ULA have maximum rate of change of arc length in broadside on position which decide resolving power of the microphone array.

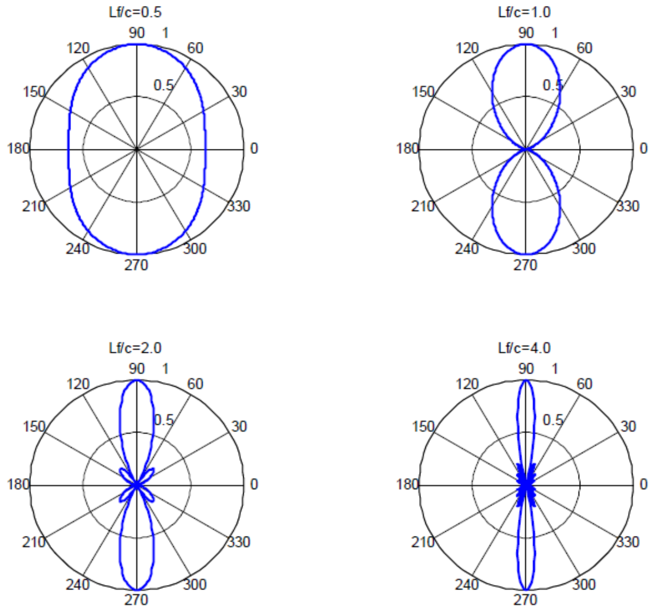

Figure 1.6. DP of the array for different parameters as specified above.

The manifold of any linear array consisting of N sensors is a curve of hyperhelical shape lying on complex N- dimensional sphere with radius $\sqrt{N}$. The Curvature of this curve depends on No. of sensors spacing between them and their lower order curvature.

Results

Shows only the noise reduction due to microphone array processing. The stationary noise suppressor in the audio stack will add 8 to 13 dB of noise reduction. The microphone array not only reduces the amount of ambient noise, but it also helps this noise suppressor to do a better job. Suppose that the signal-to-noise-ratio (SNR) in the room is 3 dB when captured with an omnidirectional microphone. With this input SNR, a stationary noise suppressor cannot do much noise reduction without introducing heavy nonlinear distortions and adding audible artifacts called musical noises. The noise reduction can add around 3 dB as well, so in this case the output SNR is 6 dB.

Under the same conditions, the microphone array reduces 13 dB of the ambient noise, and now the noise suppressor has a 16 dB SNR on its input. It can easily reduce an additional 13 dB of stationary noise without significant distortion in the signal and audible musical noises. The output SNR in this case will be 29 dB, which is 23 dB better than a system with an omnidirectional microphone. The total noise reduction of the audio stack reaches an impressive 26 dB, creating high sound quality with a very low level of distortion and artifacts.

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