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Study and Evaluation of The Uniform and Nonuniform Beam forming Systems for Reduction the Noise and Interference

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Abstract*— *The noise and interference signals are the most electric problems in the several areas in communication, for example, radar, sonar, and wireless communications. These signal are added to desired signal and caused errors in the acoustic and data. Several processing used for separate the desired signal from the noise or interference signal. One of digital signal processing technique is the beamforming system. The beamforming system (BFs) used with a microphone array with a specific form to represent a spatial filter which can extract a signal from a specific direction and reduce the signals from other directions. This paper discusses two types of beamforming systems, the first is the uniform phase and amplitude beamforming and the second is the non-uniform phase and amplitude, in this type the LMS algorithm is used to improve the signal to noise ratio.

A Matlab Simulation results prove that the non – uniform amplitude and phase array is better than a uniform array in beam steering to get approach null at interference signals and get the desired signal with less noise.

***Index Terms*— Uniform Beam forming, Adaptive Noise cancellation, Uniform Beam forming, Micro phone array.**

I. INTRODUCTION

In recent years, microphone array beamforming has received increasing attention for the achievement of speech in hands-free and distant-talker scenarios [1]. Microphone arrays can be used to reduce interference in hearing aids, teleconferencing systems, hands-free microphones in automobiles, computer terminals, speaker phones and speech recognition systems [2] [3].

As shown in fig.(1) BFs can be generally divided into two paths the first data-independent it is called fixed beamforming path because their parameters are fixed during the process. And the second data-dependent path or adaptive beamforming path always update their parameters based on the received signals [4]. In general, there are many types of beamforming system, the generalized sidelobe canceller (GSC) is a largely used to separate interference and noise from the desired signal which illustrated in fig.1, the GSC consists of three system processing units:-

1- The fixed beam former (FBF) which is designed to get desired speech signal.
2- The blocking matrix (BM) which is representing the manner of spatial rejection filter supposed to block the speech signal parts in the microphone signals. It rejects the desired signal and passes interference.
3- The interference references at its output drive a multi channel adaptive interference canceller (AIC) whose coefficients are adapted to suppress the remaining noise in the FB output.

The sound propagation from the source to the microphone is suffered from attenuation and delays. Hence, interference signals can be obtained by pair- wise subtraction of time-aligned microphone signals. [5].

II. MICROPHONE ARRAY BEAMFORMING TECHNIQUE

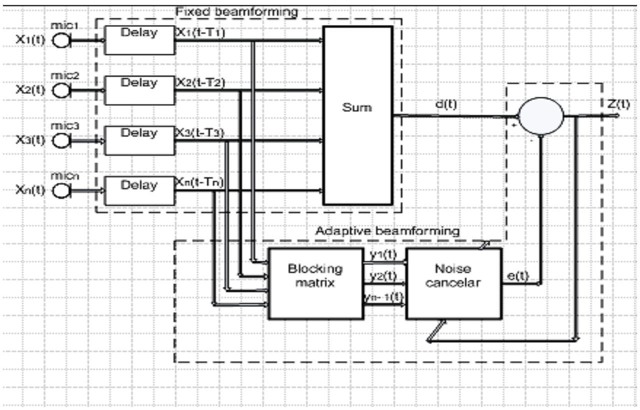

Fig.(1) Units of Generalized Side-lobe canceller system

d(t) is the output of fixed beamforming path
$y_n(t)$ is the output of blocking matrix
e(t) is the output of adaptive interference canceller
z(t)is the output of adaptive beamforming system

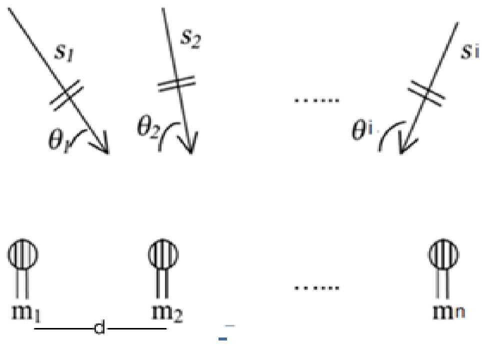

Fig.(2) Microphone array layout

The microphone array system consists of a number of microphones equal N. each microphone senses sound signals equal I, each signal of them have specific arrival angle. As shown in fig.(2)[6] each microphone in the array will thus receive all of these signals si, but it received each signal with a

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delay depends on the distance between the sources and the microphones. If the distance from microphone m1 to any microphone mn is dn. The time delay can be calculated [6]:

$$Time\ delay\ (\tau) = \frac{dn\cos\theta i}{v} \quad \dots\dots(1)$$

Where v is the velocity of sound (343 m/s) θi is the incident angle.

The signal arrival to each microphone X_n (t)

$$Xn(t) = si(t-\tau) \quad \dots\dots(2)$$

$$Xn(t) = si(t)e^{-j\omega\tau} \quad \dots\dots(3)$$

$$\omega = 2\pi fc \quad \dots\dots(4)$$

f_c is the cutoff frequency

the incoming signals are a combination of noise and ***I*** signal so it can be calculate as:

$$Xn(t) = 1/N\sum_{i=1}^{I} Amsi(t)e^{-j2\pi fcdncos\theta i/v} \quad \dots\dots(5)$$

A_m is the amplitude of each microphone.

For uniform microphone array, the delay (T_n) can calculate as:

$$Tn = \frac{dncos\beta}{v} \quad \dots\dots(6)$$

Where β is the direction of the desired signal.

While (LMS) algorithm used to calculate the amplitude and phased parameters in non-uniform beamforming. While the output of FBF can get by summation of the all microphones signals. The output of the first part of the ABF (blocking matrix) (BM) represent the difference between successive output microphone signal [6].The second parts of the adaptive path (adaptive input canceller) (AIC) contain an LMS algorithm adaptive filter which has variable parameters calculated depending on the output of fixed beam forming signal and subtracts components (the output of BM) [7].

III. MODELING OF MICROPHONE ARRAY SYSTEM

Matlab SIMULINK modeling of beamforming system is shown in Fig.(3), in this system four speech signal get by using recorder card, as shown in fig.(4) four microphones N = 4 with an inter elements distance of d = 0.057m placed in a simulated reverberant enclosure of the size (5 m) x (5m) were used. The speech sources were placed within the enclosure according to Fig.(4) with angles of θ_1 =10degree, θ_2 = 60degree, θ3 =120 andθ4 =135 (from the reference microphone).and the f_c of the filters used for speech signals are S1=700Hz, S2=2000Hz, S3=1150Hz, S1=3000Hz. These signals pass throw a low passed filter for bandwidth limitation.

Matlab function block was used to calculate the delay between each incident signal to the microphone, and the phase of the desired signal depending on the distance between the microphone and incident angle of each signal as shown fig.(5).

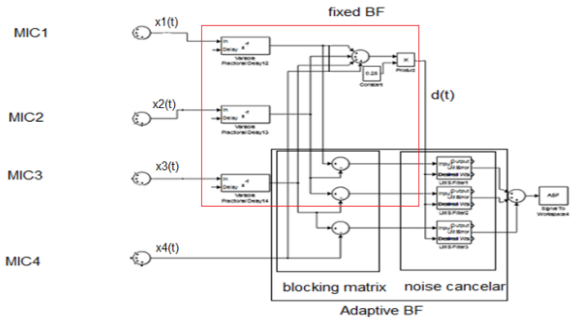

Fig.(3) general Beamforming simulation

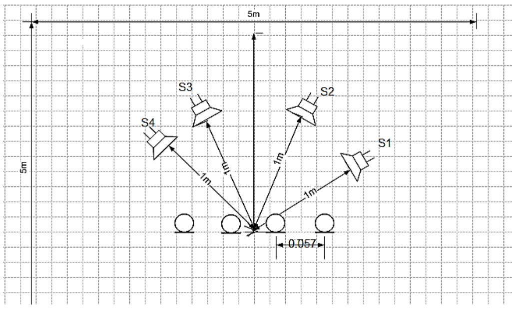

Fig.(4) The environment room

The signals in each microphone are the summation of four signals each signal shifted by (Time Delay). At first, in uniform microphone array beamforming, the signals are processed as follows:- delayed the output of each microphone by (T_n), and then the signals are summation after delay and thus we get the output of FBF. this output is the desired signal with a little interference signal, but the ABF path is started by subtracting the signal xn(t-T_n) with next signal :

$$y_n(t) = X_n(t-T_n) - X_{n+1}(t-T_{n+1}) \quad \dots\dots(7)$$

These output represent only interference signals. now each of these signals is an input to the adaptive filter, where each is compared with the signal d (t), after update parameters the output of beamforming system is sum the output of adaptive filters.

The non-uniform beamforming differs from uniform beamforming with amplitude and phase parameters. For this way construct non-uniform amplitude and phase microphone array beamforming for enhancement signal to interference ratio.

To enhance the specifications of the BF system many type of algorithm for calculation the amplitude and phase parameters can be used. This parameters added to microphone array to give a better signal to interference ratio, so this process lead to get approach null at interference signals and maximum at desired signal. In this paper LMS algorithm are used for adaptation the phase and amplitude parameters for each microphone and then tuning the result to get either one or more speech signal at the null to get best results.

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IV. SIMULATION RESULTS

A-uniform phased and amplitude beam forming

The results of the uniform phase and amplitude beam forming system are shown in fig (6 to 11) Figure 6 shows the array factor of signals at broadside. In fig.7,8,9,10 the array factor seen maximum at the desired signal and minimum at other signals.

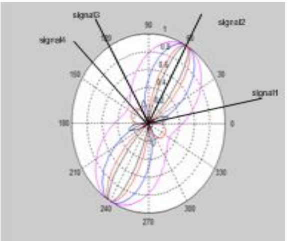

Fig.(7) array factor directed towered $\theta 1$

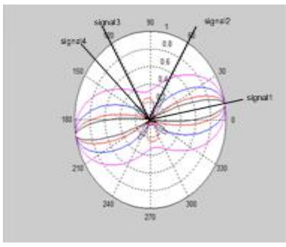

Fig.(8) array factor directed towered$\theta 2$

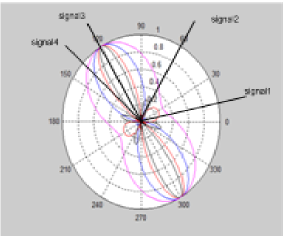

Fig.(9) array factor directed towered $\theta 3$

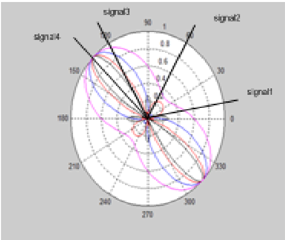

Fig.(10) array factor directed towered $\theta 4$

Fig.11 shows the output of system at each above cases. Each output is represent the desired signal with being interference signals it has the signals at other direction.

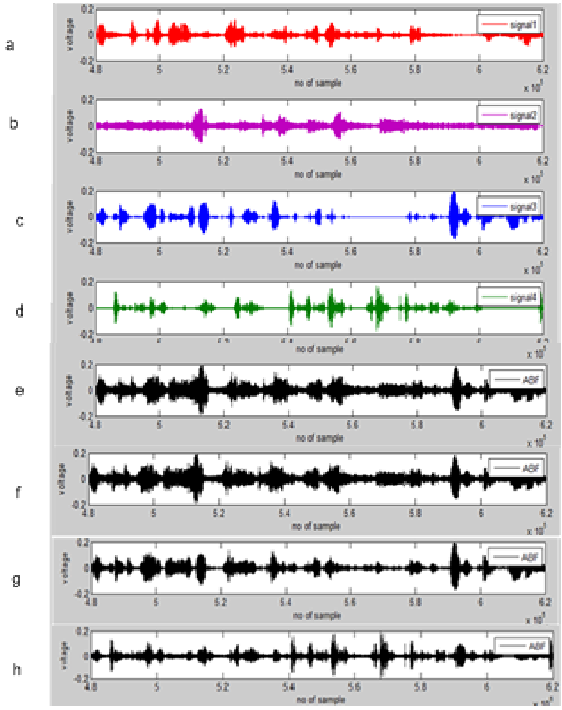

Fig.(11) input and output signals for uniform BF system

a. speech signal at theta1 b. speech signal at theta2
c. speech signal at theta3 d. speech signal at theta4
e. output of BF when the beam steer to signal1
f. output of BF when the beam steer to signal2
g. output of BF when the beam steer to signal3
h. output of BF when the beam steer to signal4

B-Nonuniform phased and amplitude beamforming

The results of the non uniform phase and amplitude beamforming system are shown in fig (12 to 16) In fig.12,13,14,15 the array factor it maximum at the desired signal and approach nulling at other signals.

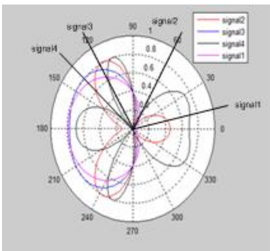

Fig.(12) array factor directed towered$\theta 1$

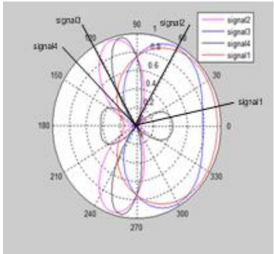

Fig.(13) array factor directed towered$\theta 2$

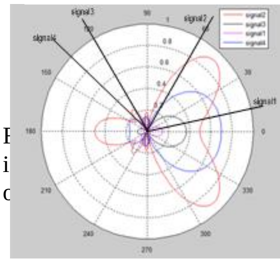

Fig.(14) array factor directed towered$\theta 3$

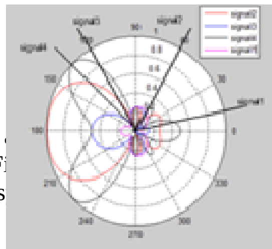

Fig.(15) array factor directed towered$\theta 4$

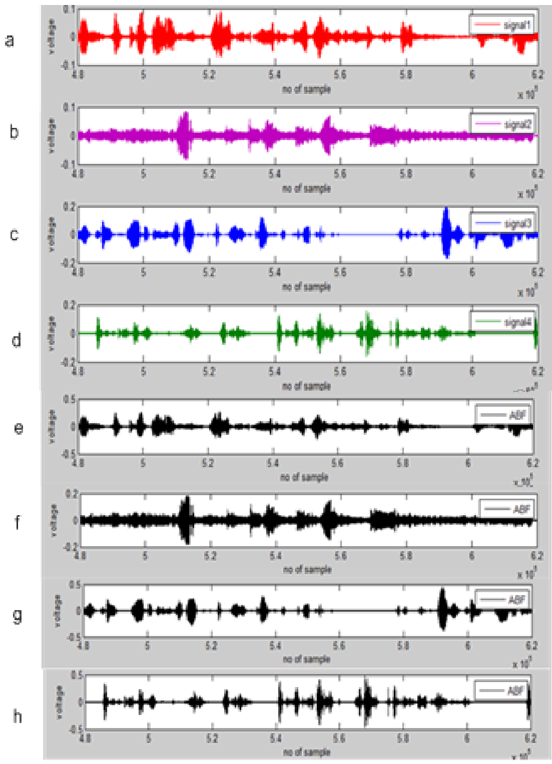

Fig.(16) input and output signals for non uniform BF system
a. speech signal at theta1 b. speech signal at theta2
c. speech signal at theta3 d. speech signal at theta4
e. output of BF when the beam steer to signal1
f. output of BF when the beam steer to signal2
g. output of BF when the beam steer to signal3

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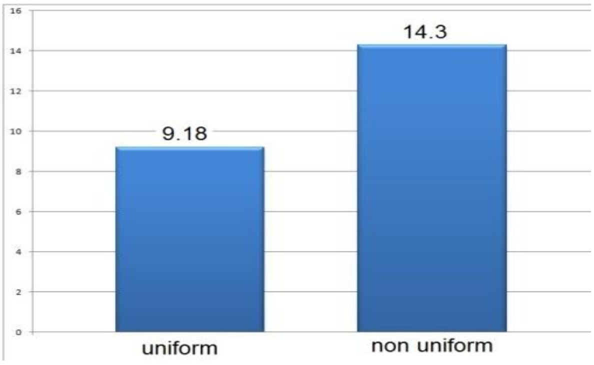

Fig 17 the difference between uniform/nonuniform systems in s/n

V. CONCLUSION

From the results, one can be observe that the use of uniform beam forming cannot get nulls, that lead to the system output contains the required signal with the interference signals that cannot be completely deleted and is clearly visible. so the unwanted signals are heard. While the non uniform beam forming system can get real nulls, the output is close to the required signal. Also the second system have S / N better than the first system but the second system more difficult and expensive because it needs algorithms used to obtain the amplitude and phase value through which we get approach null. Anyway we can say it can improve the sound signals using the beamforming technique.

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