University of Mosul
College of Electronics Engineering
Communication Department

DESIGN AND IMPLEMENTATION OF MICROPHONE ARRAY BEAMFORMING SYSTEM

Shatha M. Ali
(B.Sc. in Communication Engineering)

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Supervised By

Dr. Mahmod A. Mahmod

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By

Shatha M. Ali

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in

Communication Engineering

Supervised By

Dr. Mahmod A. Mahmod

2017 A.D. ***1438 A.H.***

Supervisor's Certification

I certify that the dissertation entitled (**Design And Implementation Of Microphone Array Beamforming System**) was prepared by **Shatha M. Ali** under my supervision at the Department of Communication Engineering, University of Mosul, as a partial requirement for the Master of Science Degree in Communication Engineering.

Signature:
Name: Dr. Mahmod A. Mahmod
Department of Communication Engineering
Date: / /2017

Report of Linguistic Reviewer

I certify that the linguistic reviewer of this dissertation was carried out by me and it is accepted linguistically and in expression.

Signature:
Name: Osama M. Mahmod
Date: : / /2017

Report of the Head of Postgraduate Studies Committee

According to the recommendations presented by the supervisor of this dissertation and the linguistic reviewer, I nominate this dissertation to be forwarded to discussion.

Signature:
Name: Prof. Dr. Khalil Hassan Sayidmarie
Date: / /2017

Report of the Head of Department

I certify that this dissertation was carried out in the Department of Communication Engineering. I nominate it to be forwarded to discussion.

Signature:
Name: Dr Younis M. Abosh
Date: / /2017

Publications:-

Some of the important results obtained in this work have appeared in the following publication**:-**

1- Mahmod A. Al zubaidy, Shatha M. Ali "Study And Evaluation Of The Uniform And Non-uniform Beamforming System For Reduction The Noise And Interference", International Journal Of Engineering And Innovative Technology, Volume 7, Issue 2, August 2017.
2- Mahmod A. Al zubaidy, Shatha M. Ali " TMS320C6713 DSP KIT Based Hardware Implementation For The Microphone Array Beamforming System" International Journal Of Computer Applications, Volume 179(6), December 2017.

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Abstract

Speech enhancement is one of the most important subjects in the signal processing area and communication. It is normally known as the restraint of additive noise, because speech signals are too non-stationary. Speech enhancement so far has had difficulties in enhancing speech or separating it from background noise and interference signals.

In this thesis, Microphone Array Beamforming (MABF) technique was used to reduce the noise and interference signal and enhance the speech signals. Three types of (MABF) system (uniform (MABF) system using two and four microphones, and non-uniform (MABF) system using four microphones) was simulated using MATLAB-SIMULINK and then implemented using TMS320C6713 KDS KIT with assistance programs. Least Mean Square (LMS) algorithm was used to calculate amplitude and phase for the non-uniform (MABF) system, each system was tested using tone signals and then tested by using speech signals.

One KIT and three PCs was required for implementation the uniform (MABF) system using two microphones, while the implementation of the other two systems requires using two KDS KITs and four PCs. The results of all simulated and implemented systems were demonstrated and discussed, the signal to noise ratio for each system were calculated and compared with other systems.

For the three systems that have been used, the signal-to-noise ratio is clearly improving. The signal-to-noise ratio of the first system is 6.18, the value of the second system is 9.18 while the latter is 14.3

Finally its clearly appeared the fact that the best-implemented system is the non-uniform (MABF) system using four microphones, follows it the uniform (MABF) system using four microphones which

have S/N ratio below the non-uniform by 5dB and the last one is the uniform (MABF) system using two microphones that have S/N ratio 3dB less than the second system.

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List of Abbreviations:-

Abbreviate	Name
ABM	**A**daptive **B**locking **M**atrix
AF	**A**rray **F**actor
AMC	**A**daptation **M**ode **C**ontrol
ADC	**A**nalog to **D**igital **C**onverter
ABF	**A**daptive **B**eam**F**orming
AIC	**A**daptive **I**nterference **C**anceller
BM	**B**locking **M**atrix
BF	**B**eam**F**orming
CCS	**C**ode **C**omposer **S**tudio
CCAFs	**C**oefficient-**C**onstrained **A**daptive **F**ilters
DAS	**D**elay **A**nd **S**um
DSP	**D**igital **S**ignal **P**rocessing
DAC	**D**igital to **A**nalog **C**onverter
FBF	**F**ixed **B**eam**F**orming
FIR	**F**ilter **I**mpulse **R**esponse
FPBF	**F**ixed **P**ath **B**eam**F**orming
FASBF	**F**ilter-**A**nd-Sum **B**eam**F**ormers
FC	**C**utoff **F**requancy
GSC	**G**eneralized **S**idelobe **C**anceller
IDE	**I**ntegrated **D**evelopment **E**nvironment
LCMV	**L**inearly **C**onstrained **M**inimum **V**ariance
LMS	**L**east **M**ean **S**equare
MABF	**M**icrophone **A**rray **B**eam**F**orming
MAX-SINR	**M**aximum **S**ignal to **I**nterference-**P**lus-**N**oise **R**atio
MVDR	**M**inimum **V**ariance **D**istortionless **R**esponse
S/N	**S**ignal to **N**oise **R**atio
TICCS	**T**ransfer **I**nformation to and from **C**ode **C**omposer **S**tudio
VAD	**V**oice **A**ctivity **D**etector

CHAPTER ONE

INTRODUCTION

1.1 General Review:-

The noise and interference signals are most electric problems in the several areas in the communication system, for example, radar, sonar, and wireless communications. These interference signals are added to the desired signal and caused errors in the acoustic and data.

The sources of interference may represent the signals other than the signal of interest, e.g. background noise or echoes. The microphones cannot distinguish between this and susceptible to noise from all directions. Even though there are some techniques available [1], and this interference cannot easily be removed from the desired source signal.

Speech Improvement is a very important and so difficult trouble for two reasons. The first is the description of the sound signal and interferences, which can be changed dramatically with time. The second reason is the performance of speech signal evaluation metrics which differ depending on the application [2].

Beamforming (BF) technique is a signal processing system which used in many applications in the communication system. This is achieved by combining elements in an array structure in such a way that signals at particular angles experience constructive interference while others experience destructive interference. This technique achieves spatial selectivity and can be applied to radio or sound waves [3]. Usually, this system reduces the unwanted signal at the output of a microphone array by means of the best (e.g., least-squares). It represents a spatial filter which can extract a signal from a specific direction and reduce the signals from other directions [1].

There are many types of beamforming systems which used to distinguish the desired signal, Generalized Sidelobe Canceller (GSC) is one of them, in this thesis (GSC) with two types of systems used to get the result, the aims of these system uses in order to improve the output signal.

1.2 Literature Review:-

Soderman, and Noble in 1975 used an end-fire microphone array with digital time delays to directionally scan the array to focus on noise sources with the intentions of rejecting background noise and reverberations in the NASA Ames (40x80) ft. wind tunnel [4].

Billingsley and Kinns in 1976 published their work on “The Acoustic Telescope”, and their telescope included a microphone array and digital computer that processed the signals and output source distributions concerning position and frequency[5].

Brooks, T. F., Marcolini M. A., and Pope D. S. in 1987 used square array frequency domain beamforming were placed in the helicopter rotor sweep area. [6].

Gramann, and Mocio in 1995 investigated the usees of adaptive beamforming vs. conventional beamforming for Aeroacoustic measurements in wind tunnels [7].

Hai Q. Dam, Sven Nordholm, Nedelko Grbi´c, and Hai H. Dam in 2003 Proposed new (ABF) employing recursively updated soft constraints for acoustic speech enhancement. The beamformer operates in a subband structure to allow a time-frequency operation for each channel consequently [8].

Zhaorong Hou and Ying Jia in 2003 used (GSC) with a new Adaptive Blocking Matrix (ABM) with exact Filter Impulse Response (FIR) structure and usual misdetection of Adaptation Mode Control (AMC) in real applications. The output of Fixed BeamForming (FBF) is used as the desired signal [9].

Alberto Abad and Javier in 2007 proposed (GSC) with (ABM) and the use of a Wiener filter [10].

Z. Qi and T. J. Moir in 2007 proposed combined three microphones Voice Activity Detector (VAD) and the noise-canceling system is studied to enhance speech recognition in an automobile environment [11].

Ernst Warsitz, Alexander Krueger and Reinhold Haeb-Umbach in 2008 illustrated new blocking matrix are proposed. It is based on a generalized eigenvalue problem whose solution provides an indirect estimation of the transfer functions from the source to the sensors [12].

Mehres Souden, Jacob Benesty, and Sofiène Affes in 2010 study the LCMV and MVDR Noise Reduction Filters in this context consists using the Linearly Constrained Minimum Variance (LCMV) and the Minimum Variance Distortionless Response beamformer (MVDR) beamformer used to reject the interference, reduce the overall mixture energy, and preserve the target signal [13].

Danilo Comminiello, Michele Scarpiniti, Raffaele Parisi Albenzio Cirillo, Mauro Falcone and Aurelio Uncini in 2011 proposed Adaptive BeamForming (ABF) technique, for speech enhancement applications, designed to be robust to no stationary interfering sources in noisy and reverberant environments [14].

Jafar Ramadhan Mohammed in 2012 proposed a simple technique for canceling the sidelobe to involve the addition of two elements, one at the end of the monopulse antenna array, which produces together a cosine pattern used for obtaining wide-angular nulls [15].

Omar W. H. in 2012 used MATLAB program and TMS320C6713DSK KIT to implementation adaptive noise cancellation for speech signal [16].

J. Tronc, P. Angeletti, N. Song, M. Haardt, J. Arendt and G. Gallinaro in 2013 review state-of-the-art techniques of beamforming in mobile satellite systems and evaluate the potential benefits/drawbacks of on-ground beamforming compared with the onboard beamforming approach [17].

S.K. Bodhe, B.G. Hogade, Shailesh Dharma Nandgaonkar in 2014 used Rectangular (MABF) Techniques for Smart Antenna [18].

Marwan Younis, Paco Lopez-Dekker and Gerhared Krieger, July 2015 used beamforming in radar system [19].

Vincent M. Tavakoli, Jasper R Jensen and Rechared Heusdens in 2017 distributed signal subspace filtering method are proposed which is not restricted to a special graph topology. Here, the Maximum Signal to Interference-Plus-Noise Ratio (MAX-SINR) criterion is used with the primal-dual method of multipliers for distributed filtering [20].

1.3 The Aims Of Project:-

The main aims of the proposed project can be summarized as:

a. Study the performance of adaptive beamformer.

b. An implementation of a simulated the (MABF) system by using MATLAB.
c. An implementation of the microphone array beamforming systems are using a Digital Signal Processing (DSP) kit with the Code Compression Studio (CCS) program.
d. Comparison between MATLAB-SIMULINK results and practical system results.

1.4 Research Methodology:-

The thesis can be divided into many steps as in following:

a. Study and analysis: In this step, a widespread reviewing process will be achieved in order to cover the subject.

Study the microphone array is one of these important subjects, within studies the sound characteristics.

b. Implementation: this step will be divided into two parts:

   The first part is the software simulation, where the MATLAB-SIMULINK software will be used.

   The second part is the hardware, where a special (DSP) kit will be used.

   This step applies to the following systems

   1. 2 microphones and 2 sources system.
   2. 2 microphones and 4 sources system.
   3. 4 microphones and 4 sources system.

c. Compare the results and Conclusion.

1.5 Dissertation Layout:-

Chapter one: It gives a brief introduction, literature review and the aim of the dissertation.

Chapter two: Illustrating the theoretical background of microphone array beamforming.

Chapter three: Software design of microphone array beamforming using MATLAB program.

Chapter four: Hardware implementation using TMS320C6713 KDS KIT

Chapter five: It contains the Conclusions and future work suggestions

CHAPTER TWO

THEORY OF THE MICROPHONE ARRAY BEAMFORMING SYSTEMS

2.1 Sound Signal Generation And Spread:-

The vibrations of the surface that is in touch with a flexible medium generate the traveling wave; this wave is a sound signal. The vibrations of the plane surface cause the layer of molecules of the medium close to the surface to compress and expand alternately.

These compressions and expansions are then transferred to the next layer of molecules and so on. At any point intimate, the space surrounding the vibrating plane will consist of waves of compressed or expanded molecules of the medium. Space which has traveling sound in it is called a sound field. The compressions and expansions of the medium at any point cause the pressure at that point to keep changing instantaneously. At any point, the variable pressure in the medium is basis the sound signal.

If the pressure variation in a purely sinusoidal manner, a single tone is heard. The sound is then said to have a single frequency. For pure sinusoidal sound, the distance between successive crests or troughs of the sinusoid is called the wavelength. The wavelength is the distance traveled by the sound signal during one cycle of the sinusoid. For any signal can calculate the wavelength by using next equation.

$$\boldsymbol{\lambda} = \frac{v}{f} \tag{2.1}$$

λ: Is the wavelength in m

v: Is the velocity of sound in m/s.

f: Is the frequency of the signal in Hz.

In general, the velocity of sound is a function of the characteristics of the medium such as its density, temperature, and steady-state pressure. Generally, the sound is slowest in the air and fastest in solids. At 20°C and at normal atmospheric pressure of 101 kPa, the sound has a velocity of 344 ms^{-1}in air [14].

Another important property of sound is the amplitude of the signal. For single tone, this is the maximum change in pressure from the steady-state value. The sound signals consist of a sum of sinusoids of changeable frequencies, amplitudes, and phases .

A single tone sound wave that is propagated only in a single direction. This direction can be taken in the positive direction of the x-axis. Such a wave is called a plane wave because if we join all the points of equal pressure in the wave, we will get a plane. Strictly speaking, plane waves can be generated only in controlled environments like narrow tubes and even then only as an approximation. Most real waves are spherical waves where the sound waves emanate in all directions from the source. By joining all the points of equal pressure for such a wave, we get a sphere. A small section of a spherical wave that has propagated for a sufficient distance can be approximated as a plane wave because the curvature of the wavefront can be approximated by a plane. The wave equation for such a plane sound wave can be written as [21].

$$\boldsymbol{p(x,t) = p_0 \cos (wt - kx)} \qquad (2.2)$$

P (x, t): The sound pressure has been expressed as a function of both spatial locations, x and time t.

p_0: Is the amplitude of the wave.

ω: Is the radial frequency (2πf).

K: Is the propagation constant (also called wave number) given by.

$$\boldsymbol{k} = \frac{\boldsymbol{w}}{\boldsymbol{v}} \tag{2.3}$$

From Equ. (2.1) and (2.3) the relationship between the propagation constant and the wavelength can be expressed as [1].

$$\boldsymbol{\lambda} = \frac{\boldsymbol{2\pi}}{\boldsymbol{k}} \tag{2.4}$$

2.2 Analysis Of Microphone Array Beamforming:-

In recent years, (MABF) has received increasing attention for the achievement of speech in hands-free and distant-talker scenarios [22]. 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 [23] and [24].

The (BF) system can generally be divided into two paths the first data-independent it is called (FBF) path because their parameters are fixed during the process. And the second data-dependent path or (ABF) paths always update their parameters based on the received signals [21].

A very famous and robust beamforming method is the generalized sidelobe canceller (GSC) which consists of three signal processing units, see Figure (2.1) the (FBF) which is designed to get a desired speech signal, the Blocking Matrix (BM) is 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, and the 3rd unit is the interference references at its output to drive a multi-channel Adaptive Interference Canceller (AIC) whose coefficients are adapted to suppress the remaining noise in the FBF output [25].

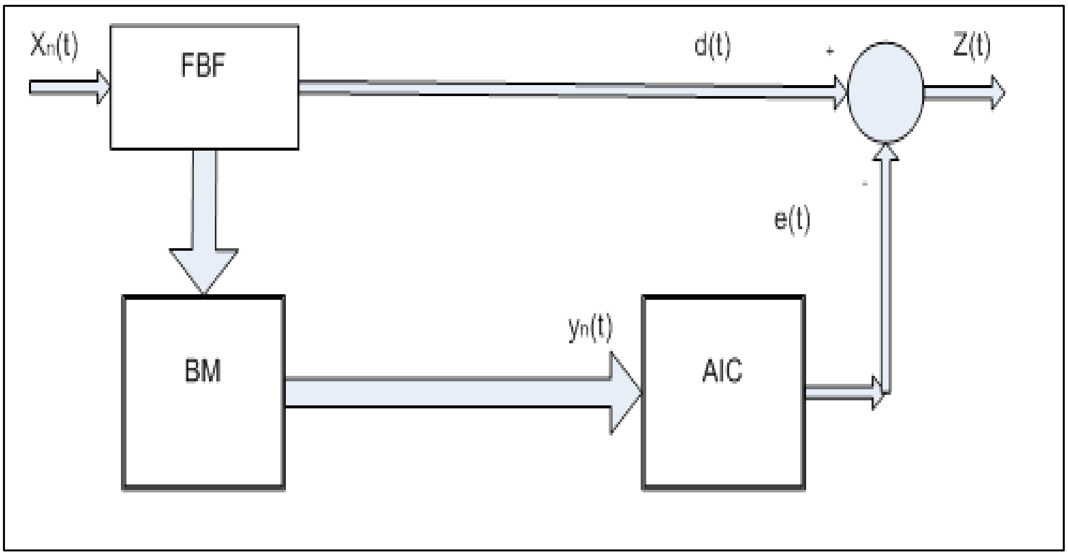


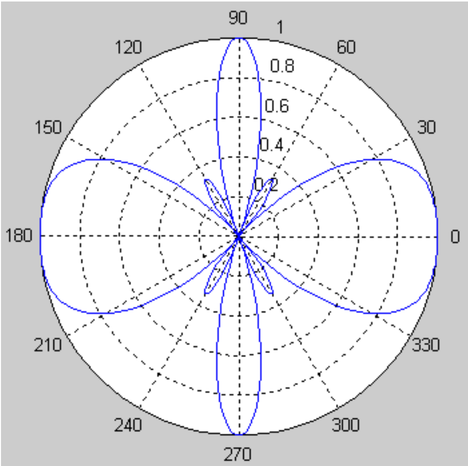
Figure (2.1): Generalized Sideloop Canceller (GSC)

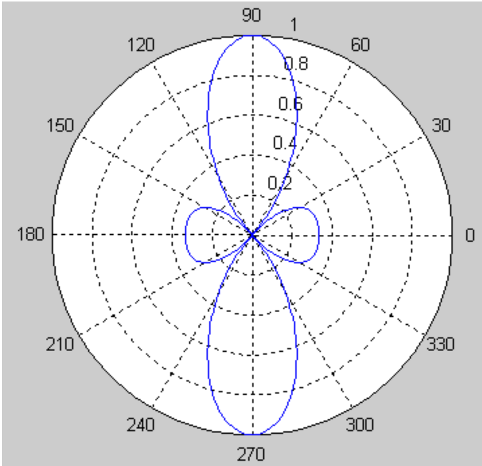
2.2.1 Spatial Aliasing:-

Spatial sampling can produce aliasing problems in a similar mode in a temporal sampling of continuous-time signals [26]. To prevent spatial aliasing effect in linear arrays, the spatial sampling theorem must be followed, which states that if λmin is the minimum wavelength of signals and d is distance between the microphone, then d must be less than the half of the minimum wavelength to avoid aliasing

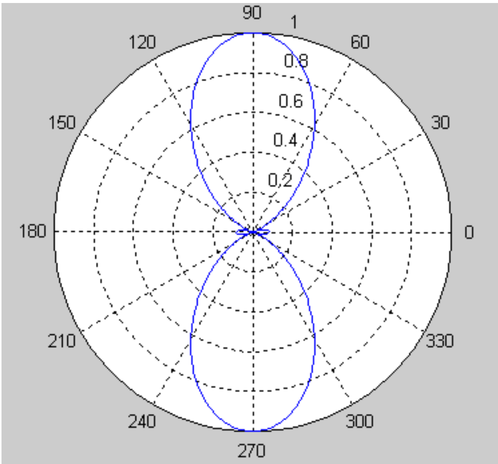
$$d \leq \lambda min/2 \tag{2.5}$$

Figure (2.2) shown the array factor for two elements linear array in three cases ($d<\lambda/2$, $d = \lambda/2$, $d > \lambda/2$) respectively.





d > λ_{min}/2 **d= λ_{min}/2**



d < λ_{min}/2

Figure (2.2): The relationship between the distance of microphones and wavelength of signal

2.2.2 Linear Microphone Array:-

Figure (2.3) shows the layout of a linear microphone array consisting of N microphones with i incident signals from different arrival angles. Each of the microphones in this array will receive a copy of each of signal s_i at a different delayed time [27].

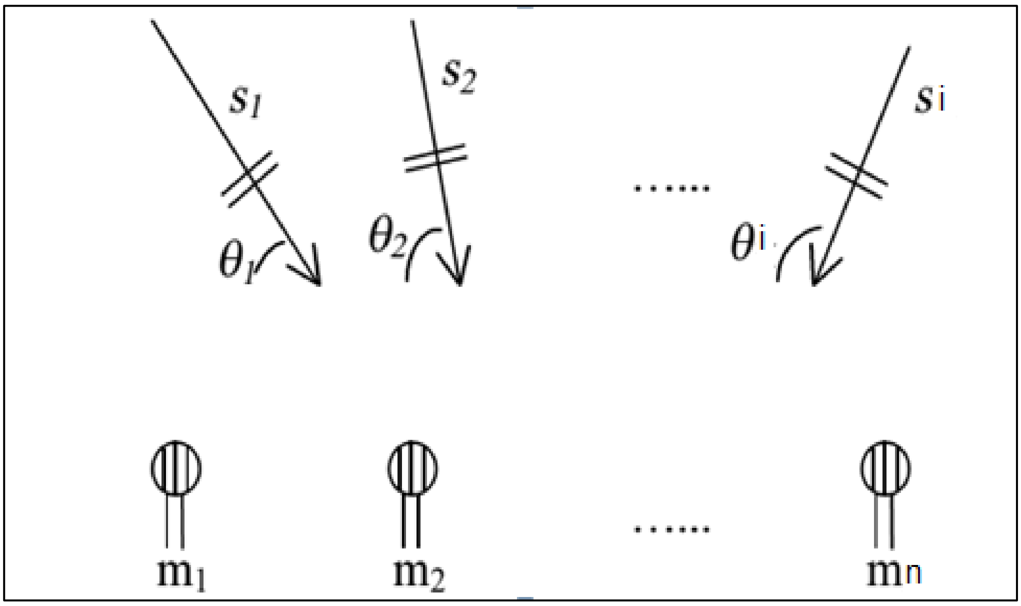


Figure (2.3): Microphone array layout [26]

The distance of the planar sound travels from each source s_i to each of the microphones m_i relative to the distance to microphone m_1 will be $d_n cos\Theta_i$. The corresponding time delays τ_n to each of the microphones m_n [27].

$$\boldsymbol{\tau_n = (dn\ cos\theta i)/v} \qquad (2.6)$$

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

$$\boldsymbol{xn(k) = \sum_1^I si(t - \tau n)} \tag{2.7}$$

$$\boldsymbol{xn(k) = \sum_1^I si(k) e^{-jkdncos\theta i}} \tag{2.8}$$

The Equ. (2.7) illuminates all signals input to each microphone.

2.2.3 Fixed Path Beamforming (FPBF) Techniques:-

The fixed beamforming (FBF) is used to control the direction of the main beam (Beam Steering).

Figure (2.4) Shows the typical (FBF) system where d (k) is the output signal of the sample index k, and x_n (k) is the output signal of nth microphone (n=0 …N) [23].

There are three types of (FPBF) techniques [10].

2.2.3.1 Delay And Sum Beamforming (DASBF):-

The famous technique to generate the output of an array is the delay and sum technique. The output of the array is written as [27]:

$$\boldsymbol{d(k) = \sum_{n=1}^N W_n a_n X_n(t - T_n)} \tag{2.9}$$

The weighting factors w_n for all sensors will be $1/N$; this allows an 'averaging' such that the output has nearly the same amplitude as the input, and delays T_n in Equ. (2.8) are chosen to help enhance the beam shape and reduce sidelobe levels. a_n is the amplitude of each microphone. The output of this model is thus [27].

$$\boldsymbol{d(k) = \frac{1}{N} \sum_{n=1}^N a_n e^{jT_n} X_n(t)} \tag{2.10}$$

For uniform amplitude and phase the parameter a_n in Equ. (2.9) equal 1 and T_n can be calculated as:

$$\boldsymbol{T_n = Kd_n cos\beta} \tag{2.11}$$

When β is the angle of the desired signal.

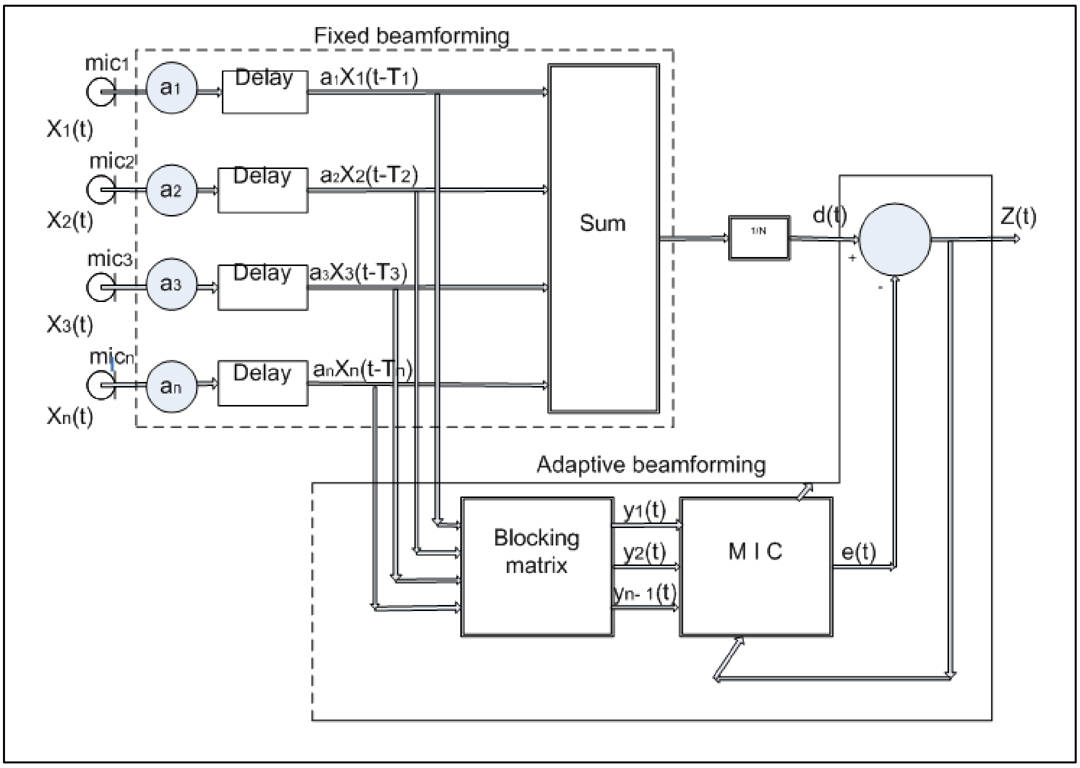


Figure (2.4): Diagram of GSC with delay and sum

To enhance the specifications of the (BF) system many types of algorithm for calculating the amplitude and phase parameters can be used. This parameter added to the microphone array to give a better signal to interference ratio, so this process leads to get either null or approach null at interference signals and maximum at the desired signal. The (LMS) algorithm is used for adapting the phase and amplitude parameters for each microphone and then tuning the result to getting either one or more speech signal at the null to get best results. Figure (2.5) shows the system that used to get the amplitude and phased parameters [28].

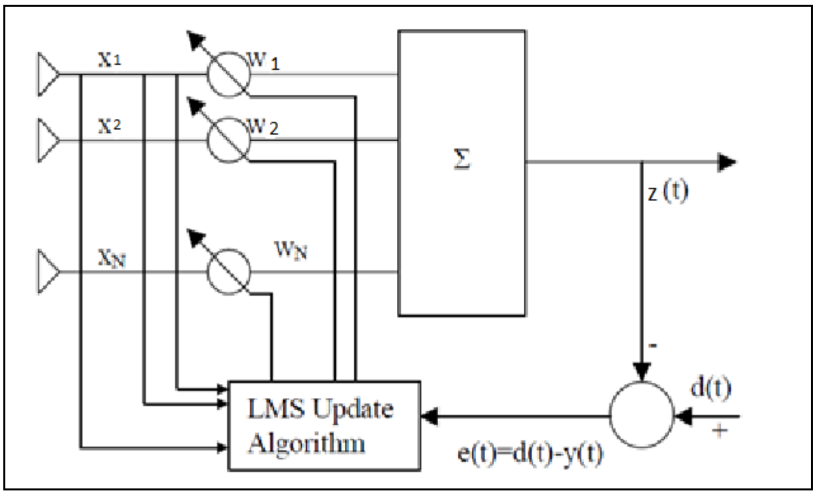


Figure (2.5): LMS adaptive beamforming network [28].

2.2.3.2 Filter-And-Sum Beamforming (FASBF):-

In (FASBF), both the amplitude and phase weights are frequency dependent. The (FASBF) can be generalized to alter-and-sum beamformer where rather than a single weight, each microphone signal connects to filter, and each signal is filtered before they are combined. The filtered channels are then summed [26], according to

$$\boldsymbol{d}(\boldsymbol{k}) = \sum_{\boldsymbol{n}=\mathbf{1}}^{\boldsymbol{N}} \boldsymbol{W}_{\boldsymbol{n}}(\boldsymbol{f})\, \boldsymbol{X}_{\boldsymbol{n}}(\boldsymbol{f}) \qquad (2.11)$$

2.2.3.3 Post-Filtering:-

To improve the performance of a (FASBF) algorithm used a post-filtering technique. A Wiener post-filter approach makes use of the information about the desired signal acquired by the spatial filtering, to achieve additional frequency filtering of the signal [29]. It makes use of cross-spectral density functions between channels, which improves the beamformer cancellation of noise.

2.2.4 Adaptive Path:-

The adaptive path can be divided into two parts. The first part is blocking matrix (BM), which is a kind of spatial rejection filter as shown in figure (2.4). It rejects the desired signal and passes the interference [23]. In this thesis used two types of (BM), the output of the 1st type is the differences between successive signal samples. If the signals when the input to the (BM) stage are absolutely in phase, then this works can be found [27].

$$\boldsymbol{yn}(\boldsymbol{t}) = \boldsymbol{Xn}(\boldsymbol{t} - \boldsymbol{Tn}) - \boldsymbol{Xn} + \mathbf{1}(\boldsymbol{t} - \boldsymbol{Tn} + \mathbf{1}) \tag{2.12}$$

The 2nd type can be a variable blocking matrix with a Coefficient-Constrained Adaptive Filters (CCAFs) by using adaptive filters to compare the output of the microphone.The (CCAFs) in the blocking matrix minimize the output signals of the (BM). Because of the constraints in the (CCAFs), this minimization leads to the minimization of the target signal leakage at the (BM) output. The minimization varies the spatial filtering pattern of the (BM) according to the target direction, resulting in target tracking [10].

The second part of the adaptive path is an (AIC) contain an (LMS) algorithm adaptive filter which has variable parameters calculated depending on the output of (FBF) signal and subtracts components (the output of BM) [23].

2.2.5 Adaptive Filter:-

The adaptive filter is the heart of the (AIC) blocks, which is a computation device that attempts to model and compare two signals.

The adaptive filter is defined by four aspects:

1. The signals being processed by the filter

2. Error signal shows the differences between the output signal of the filter and the input signal

3. When compared input-output relationship can get the parameters then can be changed to alter the filters.

4. The adaptation algorithm should be used for adaptive filter [30, 31, 32 and 33].

Adaptive filters are used in many applications, such as a noise canceller, an echo canceller, an adaptive equalizer. The necessity of their implementations is growing up in many fields. Adaptive filters need different performances of lower power dissipation; the speed is high and good convergence properties [34].

The structure of the (LMS) adaptive filter is shown in Figure (2.6) [35]. The filter input signal s (k) feds into the delay line and shifted every sampling instance. The taps of the delay line provide the delayed input signal matching the depth of delay elements.

The tap outputs are multiplied by the corresponding coefficients, the output of the (LMS) adaptive filter is the sum of these products. The error signal is defined as the difference between the most wanted signal and the filter output signal. The tap coefficients are updated by the comparison between the input signals and the scaled error signal [35].

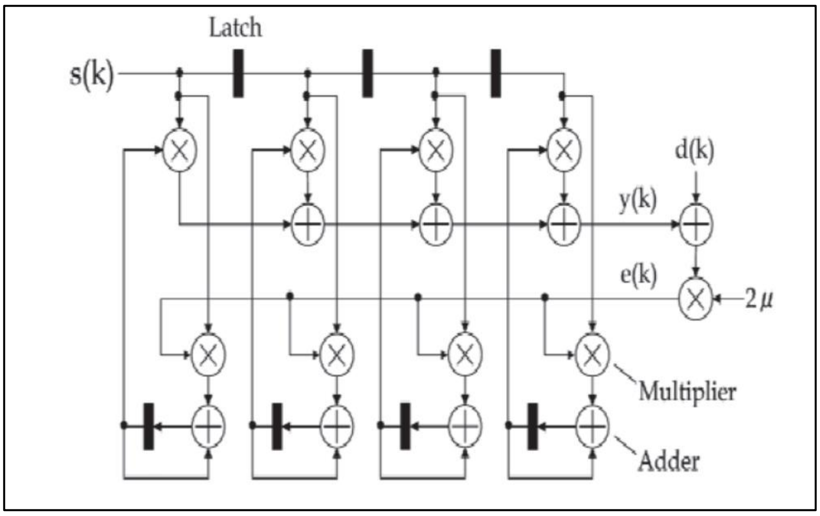


Figure (2.6): Structure of the 4-tap LMS adaptive filter [35]

$$\boldsymbol{S}(\boldsymbol{k}) = [\boldsymbol{s}(\boldsymbol{k}), \boldsymbol{s}(\boldsymbol{k}-\mathbf{1}), \dots, \boldsymbol{s}(\boldsymbol{k}-\boldsymbol{N}+\mathbf{1})]^{\boldsymbol{T}} \quad (2.13)$$

S (k): It is an input signal

K: It means time instance

T : It is a transpose of the vector.

The output signal of this system is represented.

$$\boldsymbol{y}(\boldsymbol{k}) = \boldsymbol{w}(\boldsymbol{K})\boldsymbol{S}(\boldsymbol{k})^{\boldsymbol{T}} \quad (2.14)$$

$$\boldsymbol{w}(\boldsymbol{k}) = [\boldsymbol{w}_{\mathbf{0}}(\boldsymbol{k}), \boldsymbol{w}_{\mathbf{1}}(\boldsymbol{k}), \dots, \boldsymbol{w}_{\boldsymbol{N}-\mathbf{1}}(\boldsymbol{k})]^{\boldsymbol{T}} \quad (2.15)$$

W (k): is the N-th coefficient vector represented as

$$\boldsymbol{W(k+1) = W(k)\ 2\mu e\ (k)\ S\ (k)}$$

(2.16)

The step-size parameter determines the convergence speed and the accuracy of the estimation.

The error signal is obtained by [35].

$$\boldsymbol{e}(\boldsymbol{k}) = \boldsymbol{d}(\boldsymbol{k}) - \boldsymbol{S}(\boldsymbol{k}) \quad (2.17)$$

CHAPTER THREE
MATLAB SIMULATION FOR THE MICROPHONE ARRAY BEAMFORMING SYSTEMS

3.1 Why Using MATLAB:-

In computer programs, there are many programming languages, and one of these languages is the MATLAB. This program is one of the most efficient programs in computer systems.

It integrates computation, visualization, and programming environment. MATLAB is a modern programming language environment, it has complicated data structures, contains built-in editing, debugging tools and supports object-oriented programming. All these qualities make MATLAB an excellent tool for teaching and research [36].

When this program compared with other computer languages (e.g. C, FORTRAN) there are many advantages for solving technical problems. The software package has been commercially accessible since 1984 and is now considered as a standard tool at most universities and industries worldwide.

In this chapter, the MATLAB-SIMULINK program was used to represent the (MABF) system with its two types; the uniform (phase and amplitude MABF) and the non-uniform (phase and amplitude MABF), each type was tested with two types of signals, with single tone signals as an inputs and with the real acoustic speech signals as an input signal.

3.2 Microphone Array And System Modeling:-

The model of two types of microphone array was implemented using MATLAB-SIMULINK, 2 microphone array and 4 microphone array; each system was tested with a single tone and acoustic speech signals.

3.2.1 Uniform Microphone Array Beamforming System Using Two Microphones And Two Sources:-

In this system, a linear array of N = 2 MICs was used with an inter-element distance of d = 0.085m placed in a simulated reverberant enclosure of the size (5 m) x (5m). The speech sources were placed within the enclosure as shown in figure (3.1) with angles of θ1 = 60 degrees, θ_2 = 120 degrees and the fc of the sine wave signals are S_1=2000Hz and S_2=1150Hz.

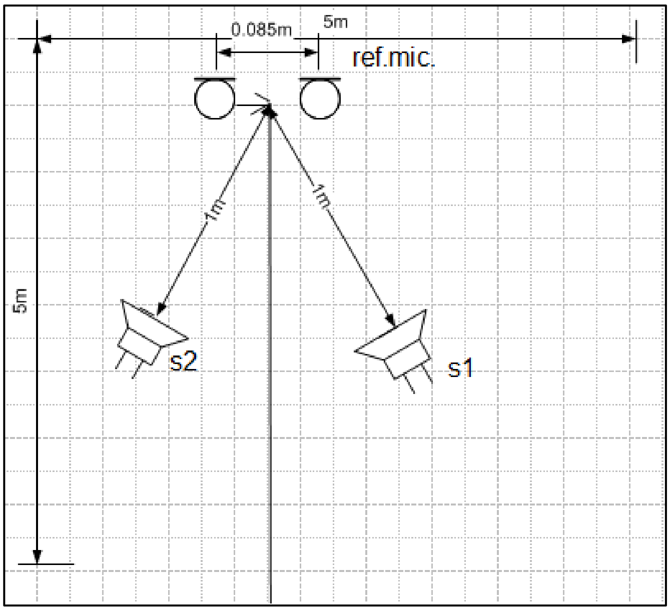
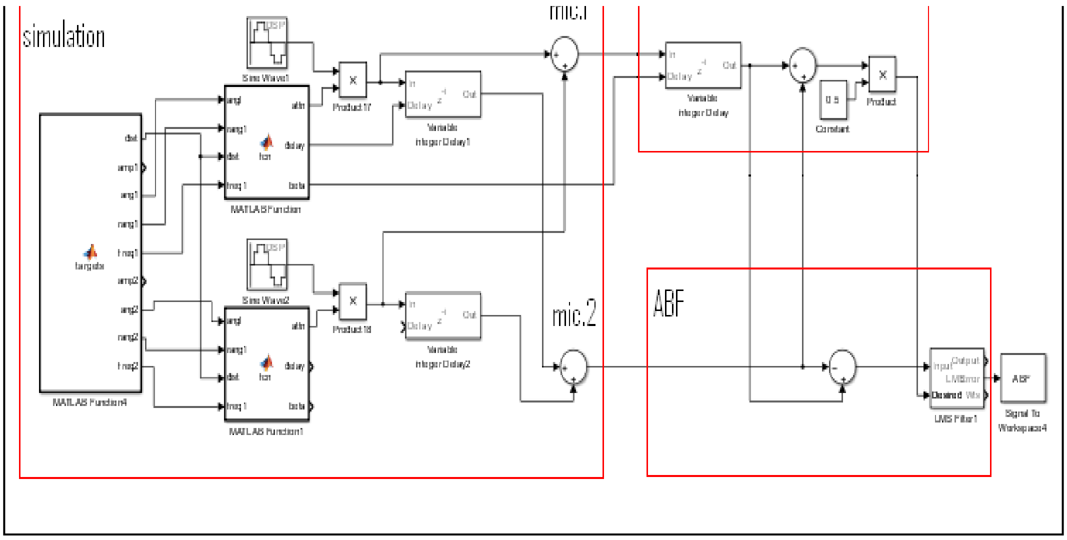


Figure (3.1): 2 MICs, 2 sources environment room

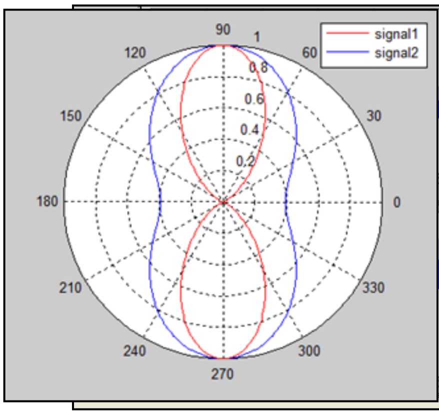
The MATLAB-SIMULINK circuit for the (MABF) system is shown in figure (3.2); one can see that the circuit consists of two parts, the environment room, and the (BF) part.

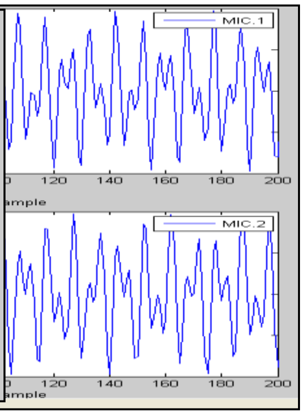
The environment room part consists of many units such as a parameter definition block, two blocks to calculate (attenuation, delay, and desired signal angle), two single tone signal generators, and the array factor calculation units. While the (BF) part consists of two subparts, the (FBF) part (which consist of summation and delay blocks) and the (ABF) part (which consist of subtracting and adaptive filter blocks).

Figure (3.2): Simulation system for 2 MICs, 2 sources tested by single tone signals



After feed, the parameters definition block with the parameters stated above (figure) 3.1, the Array Factor (AF) was calculated by the array factor calculation units as shown in figure (3.3) and the input signals to the two microphones are shown in figure (3.4).





): The array factor for d signal2 at broadside (2 MICs, 2 sources)

figure (3.4): The input tone signals to each microphone in case (2 MICs, 2 sources)

As well as the first microphone assumed as a reference, his output will be the summation of the two signals without delay, while the output of the second microphone must be the summation of the two signals but each signal with delay calculated by the two calculation blocks.

In the (FBF) stage, the output of the reference microphone delayed by the desired signal time delay and then added to the output of the second microphone to steer the direction of the array factor of the desired signal.

In the (ABF) stage, the delayed reference microphone signal will be subtracted from the output of the second microphone; then this output will represent one of the adaptive filter inputs. While the output of the (FBF) represents the second input of the adaptive filter, the output of the adaptive filter is the output of (MABF).

When tuning the beam steering of microphone array at θ1 = 60 degrees and then θ2= 120 degrees the (AF) of the microphone array are shown in figures (3.5) A and B, and the output of (MABF) is a sine wave with frequency equal 2000Hz and 1150Hz as shown in figures (3.6).

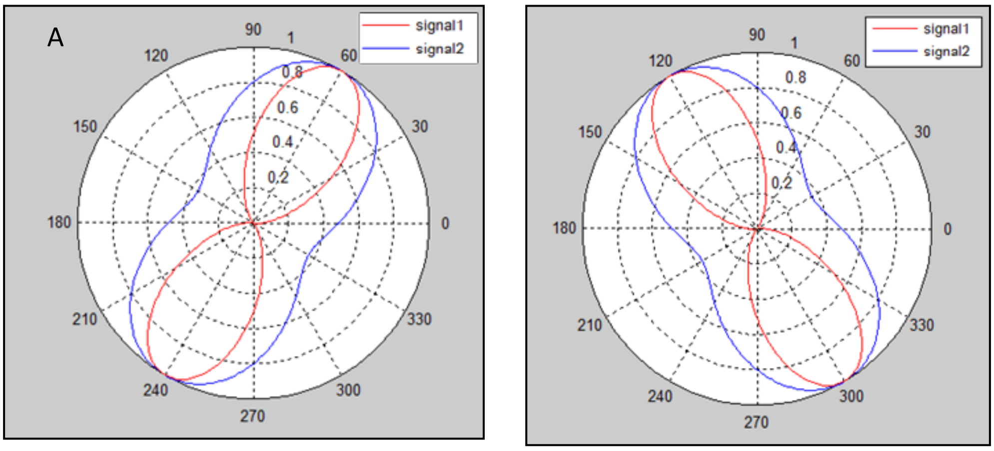


Figure (3.5): A the array factor for signal at θ_1 = 60degree
B the array factor for signal at θ_2 = 120degree in case (2 MICs, 2 sources)

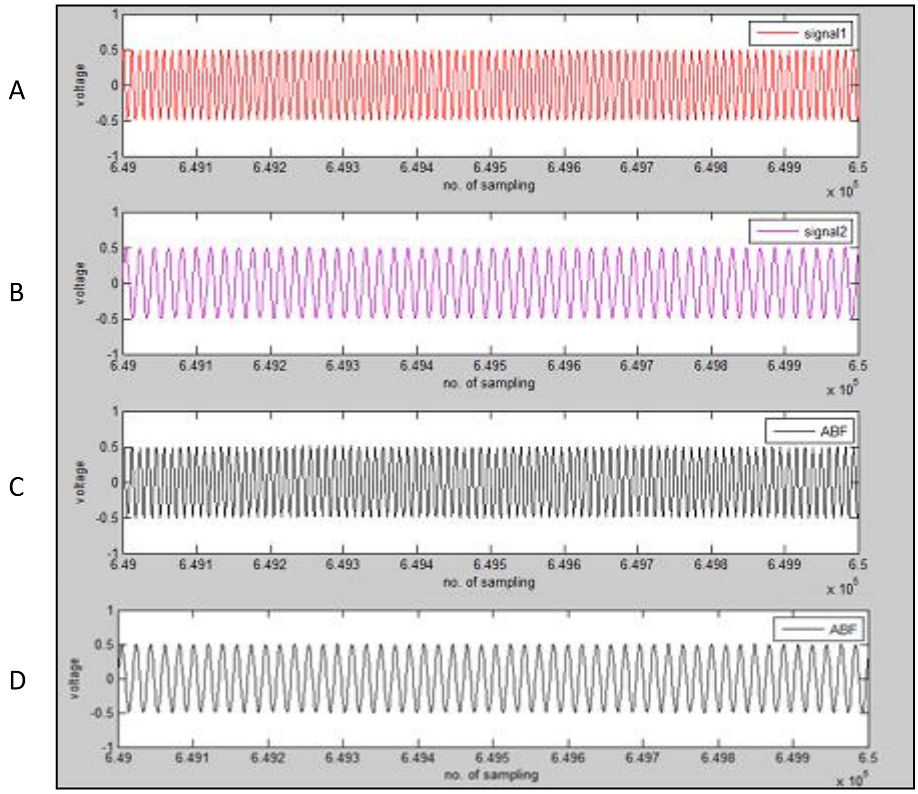


Figure (3.6): (ABF) (single tone inputs signals) & output signals
in case (2 MICs, 2 sources)
A ton signal1 B ton signal2
C (ABF) output when the beam direction set toward the signal at θ_1
D (ABF) output when the beam direction set toward the signal at θ_2

After testing the system using single tone signals, the system must be tested using recorded speech signals with a cutoff frequency equal the frequency of a sine wave signal. This system must have some changes as shown in figure (3.7).

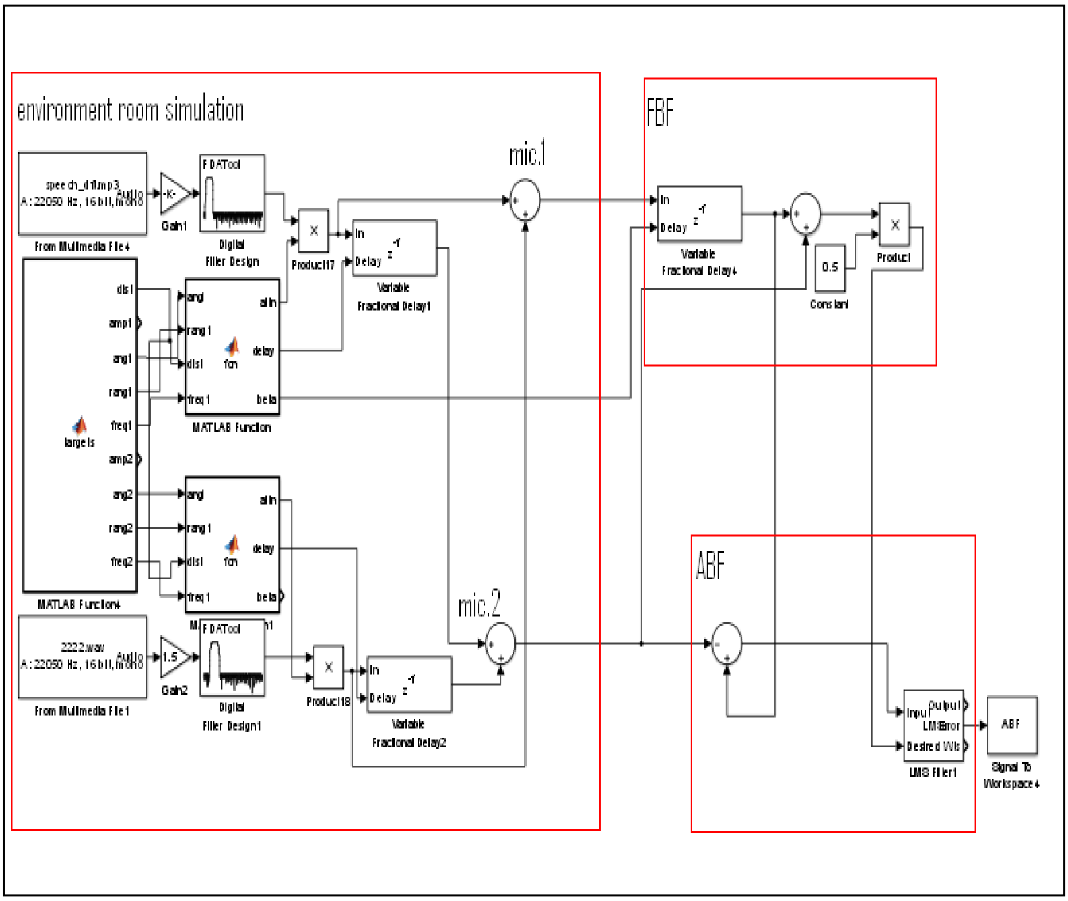


Figure (3.7): Simulation system for 2 MICs, 2 sources tested by speech signals

At the simulation system above, there is two multimedia file block will replace the two signal generators blocks and the digital filters were added to limit the bandwidth and to adjus the cutoff frequency of the signals.

Figure (3.8) shows the input speech signals to each microphone at when the array factor of two microphones at broadside.

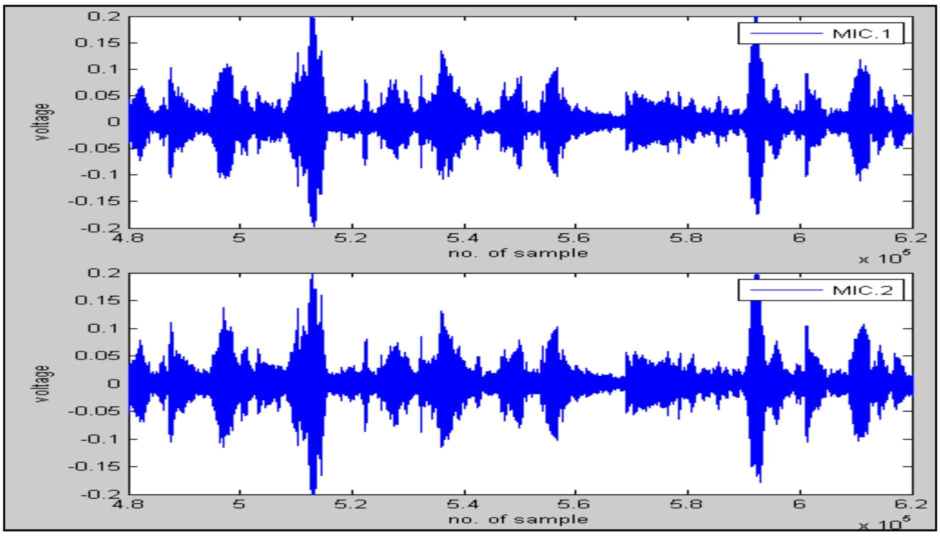


Figure (3.8): The input speech signals to each microphone in case (2 MICs, 2 sources)

Figure (3.9) shows the speech signals and adaptive beamforming output when the beam steers toward θ_1 = 60degrees and θ_2 = 120degrees respectively.

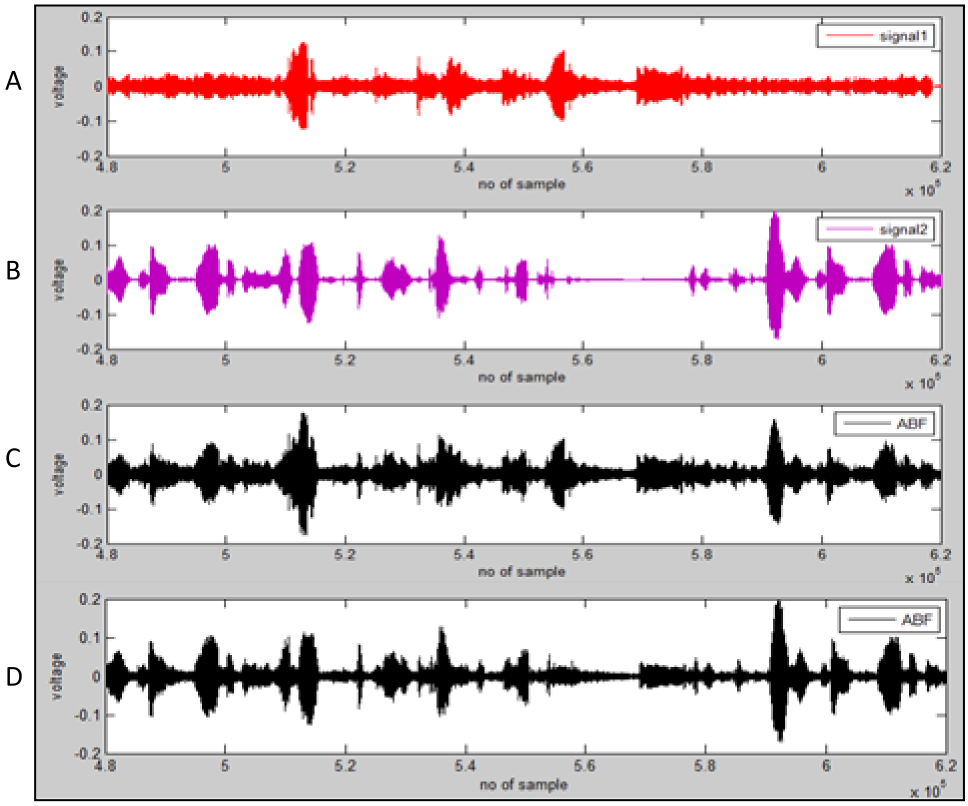


Figure (3.9): (ABF) (speech input signals) & outputs signals in case (2 MICs, 2 sources)
A speech signal1 B speech signal2
C (ABF) output when the beam direction set toward the signal at θ_1
D (ABF) output when the beam direction set toward the signal at θ_2

From figure (3.9) one can see that the output of (ABF) is not comparing the desired signal that means the output of the first stage of the system (FBF) uptake desired signal and contain some of the interfering signals and this interference signal cannot cancel through the adaptive path, because of the speech signals have a wide band of frequency, therefore, there are different results between singleton and the speech signal.

3.2.2 Uniform Microphone Array Beamforming System Using Two Microphones And Four Sources:-

In this system, a linear array of N = 2 MICs with an inter-element distance of d = 0.057m placed in a simulated reverberant enclosure of the size (5 m) x (5m) was used. The speech sources were placed in the enclosure according to figure (3.10) with angles of θ_1 =10 degrees, θ_2 = 60 degrees, θ3 = 120 degrees, and θ4=135 degrees. And the frequency of the singleton signals is S1=700Hz, S2=2000Hz, S3=1150Hz, S4=3000Hz. The simulation system for 2MICs 4sources uniform phase and amplitude are shown in figure (3.11), the (AF) for these signals shown in figure (3.12).

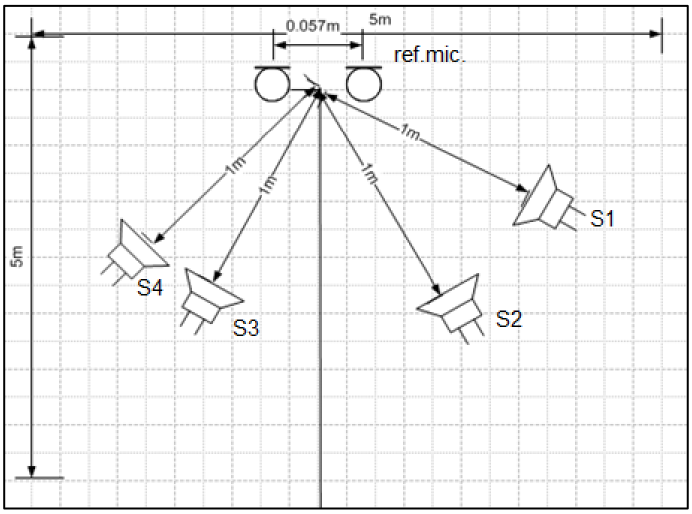


figure (3.10): 2 MICs, 4 sources environment

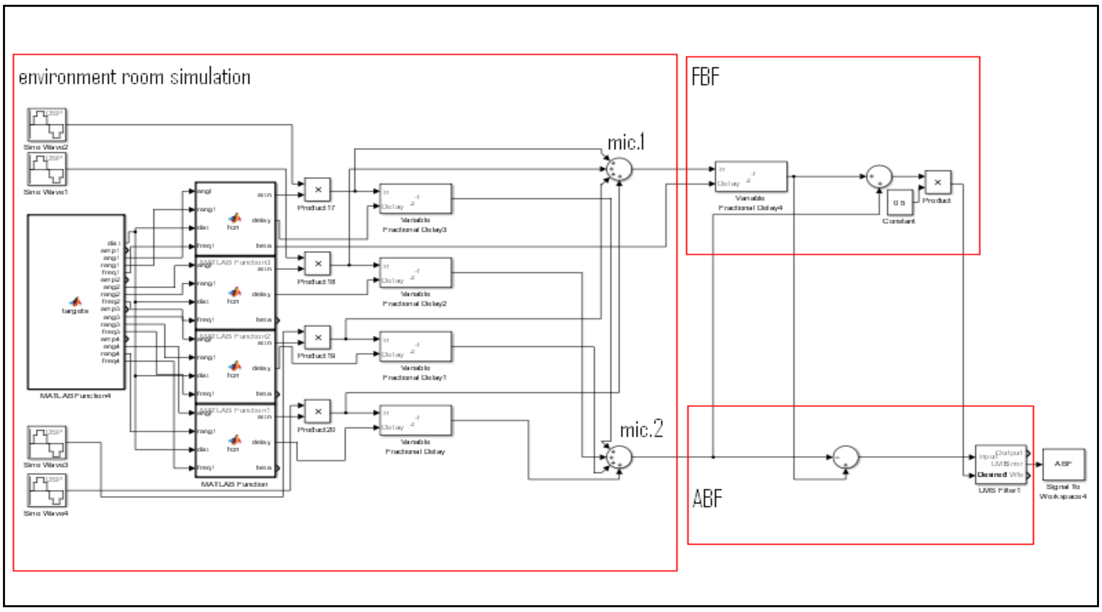


Figure (3.11): Simulation system for 2 MICs, 4 sources uniform phase and amplitude tested by single tone signals

In this system, four sine wave signal generators are used, and each signal needs a Matlab block for calculating the required parameters.

Figure (3.13) shows the signal at each microphone when the (AF) of two microphones at broadside. Figure (3.14) shows the (AF) for a signal when steering to the desired signal. Figure (3.15) shows the input signals and output signals of the system.

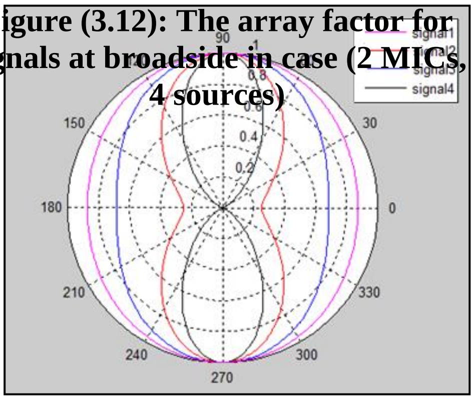


Figure (3.12): The array factor for signals at broadside in case (2 MICs, 4 sources)

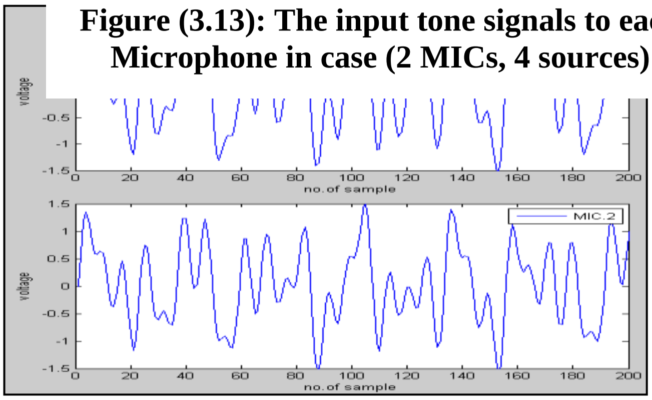
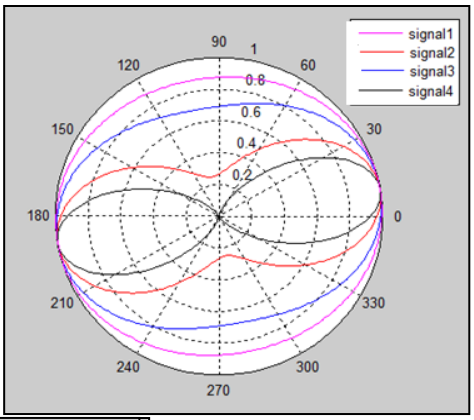
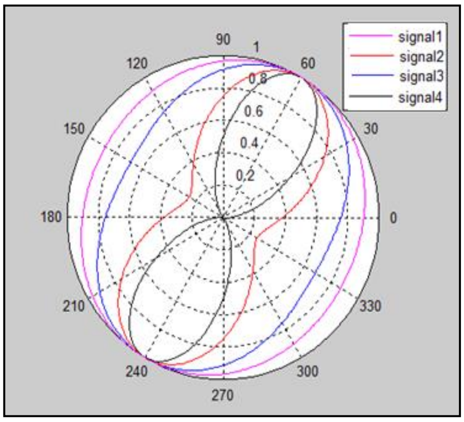
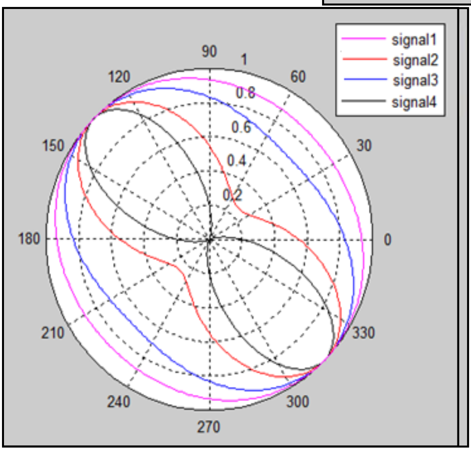


Figure (3.13): The input tone signals to each Microphone in case (2 MICs, 4 sources)

As well as the first microphone assumed as a reference, his output will be the summation of the four signals without delay, while the output of the second microphone must be the summation of the four signals but each signal with delay calculated by the four calculation blocks.







se (2 MICs, 4 sources)
at θ2 = 60degree
at θ4 = 135degree

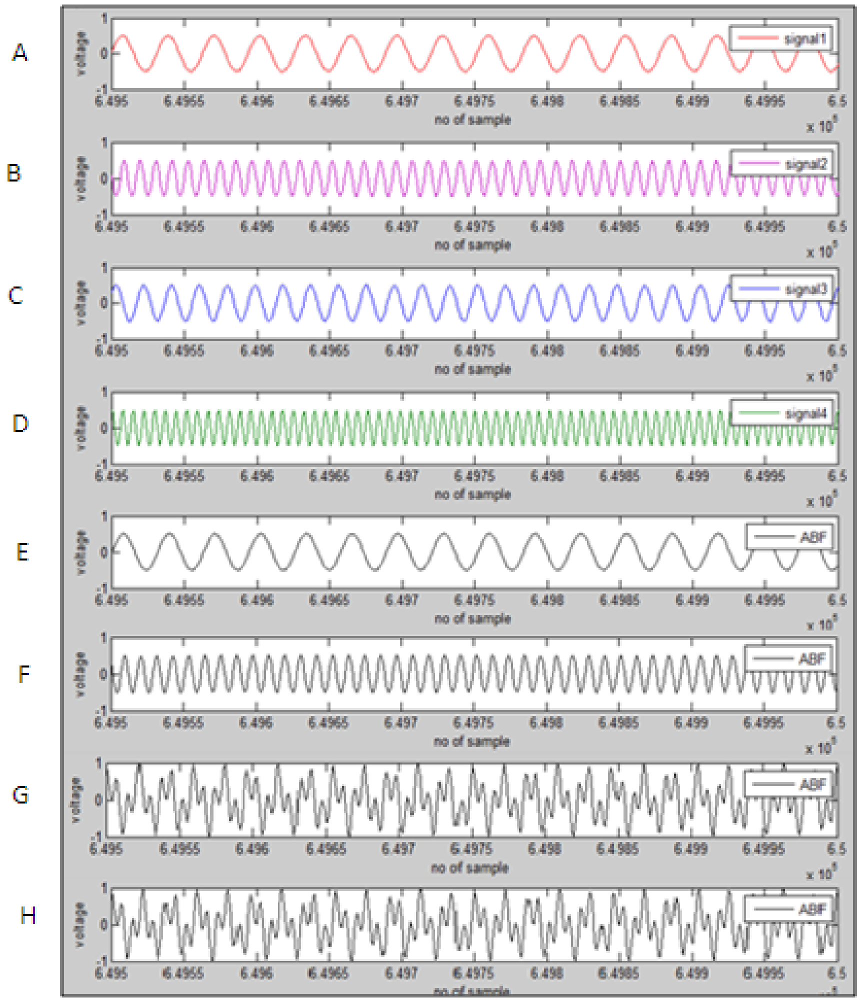


Figure (3.15): (ABF) (single tone inputs signals) & outputs signals in case (2 MICs, 4 source)
A tone signal1 B tone signal2 C tone signal3 D tone signal4
E (ABF) output when the beam direction set toward the signal at θ_1
F (ABF) output when the beam direction set toward the signal at θ_2
G (ABF) output when the beam direction set toward the signal at θ_3
H (ABF) output when the beam direction set toward the signal at θ_4

From figure (3.15) one can notice that when the beam directed toward the signal at θ1and when the beam directed toward θ2, the output of the system is a sine wave with frequency equal 700Hz and 2000Hz respectively, for two cases (ABF) systems canceled all interference signals and the output represents the desired signal. But when the beam directed toward θ3 and when the beam directed toward θ4, the output of the systems is mixed between the desired signal and the interference signals, in this case, the system cannot delete all interference signals.

When using speech signals as a test system, figure (3.16) shows the simulation system.

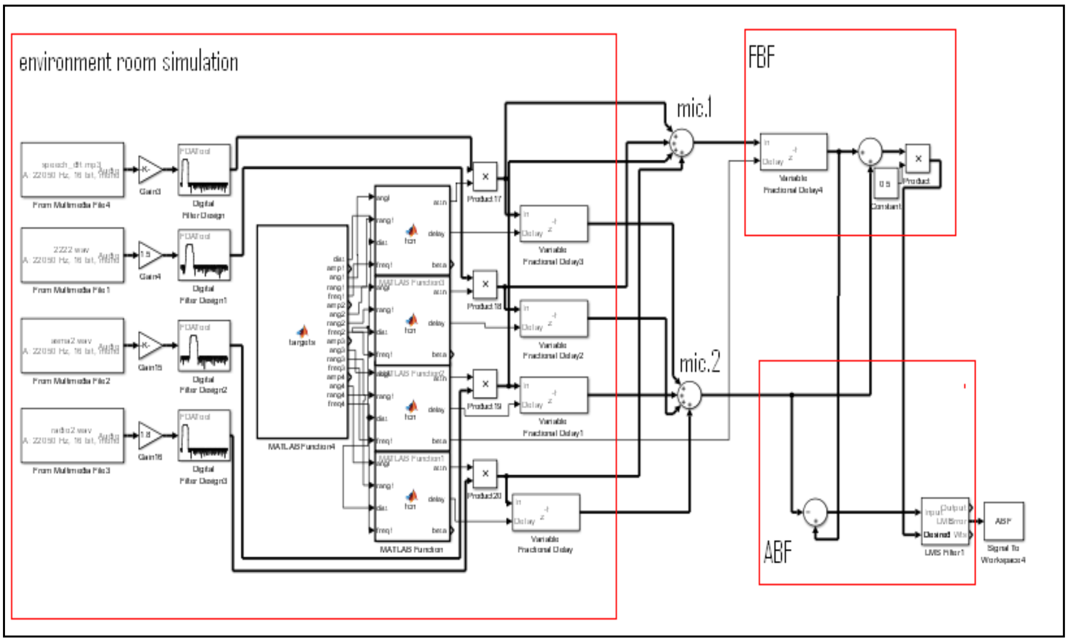


Figure (3.16): Simulation system for 2 MICs, 4 speech sources uniform phase and amplitude

In the simulation system above, there are four multimedia file blocks will replace the four signal generators blocks and four digital filters were added to limit the bandwidth and to adjus the cutoff frequency of these signals.

Figure (3.17) represents the signal at each microphone. The output of ABF system can notice in figure (3.18).

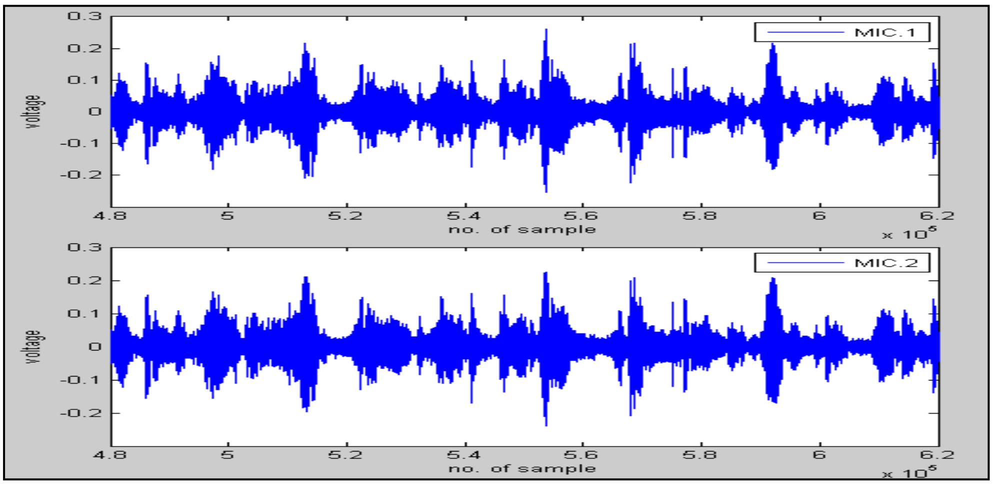


Figure (3.17): The input speech signals to each microphone in case (2 MICs, 4 sources)

A
B
C
D
E
F
G
H

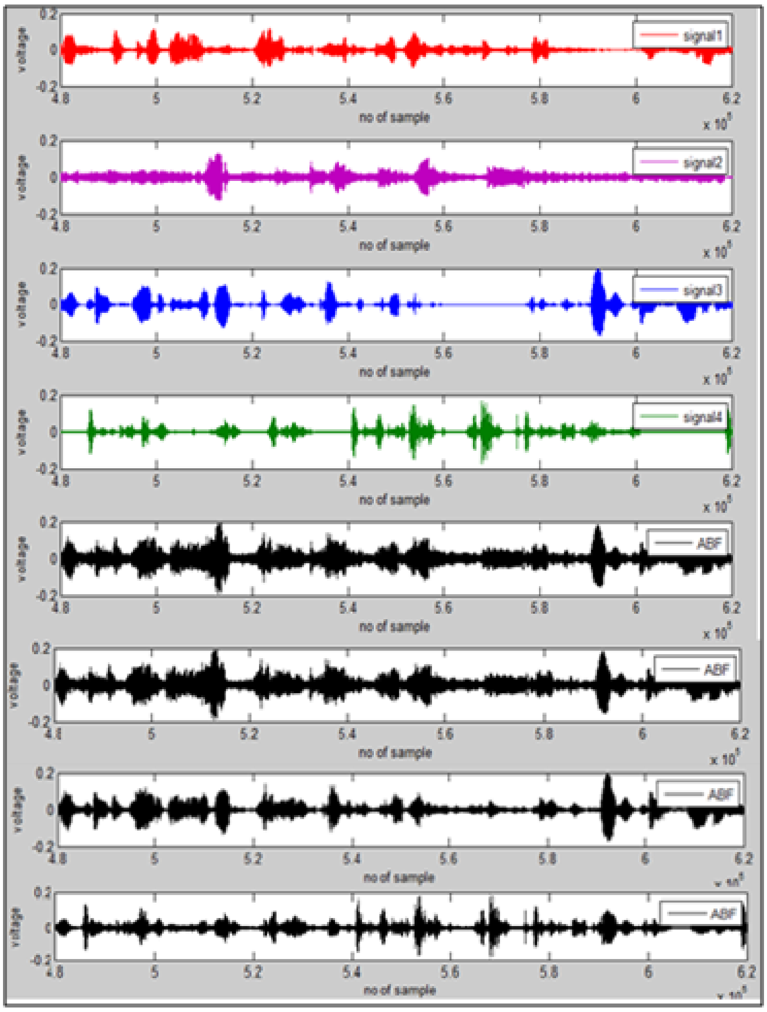


Figure (3.18): (ABF) (speech inputs signals) & outputs signals in case (2 MICs, 4sources)
A speech signal1 B speech signal2 C speech signal3 D speech signal4
E (ABF) output when the beam direction set toward the signal at θ_1
F (ABF) output when the beam direction set toward the signal at θ_2
G (ABF) output when the beam direction set toward the signal at θ_3
H (ABF) output when the beam direction set toward the signal at θ_4

3.2.3 Microphone Array Beamforming System Using Four Microphones and Four Sources:-

In this system, a linear array of N = 4 microphones with an interelement distance of d = 0.057m placed in a simulated reverberant enclosure of the size (5 m) x (5m) was used. The speech sources were placed within the enclosure according to figure (3.19) with angles of θ1 =10 degrees, θ2 = 60degrees, θ3 = 120 degrees and θ4 = 135 degrees. And the fc of the speech signals is S1=700Hz, S2=2000Hz, S3=1150Hz, S1=3000Hz.

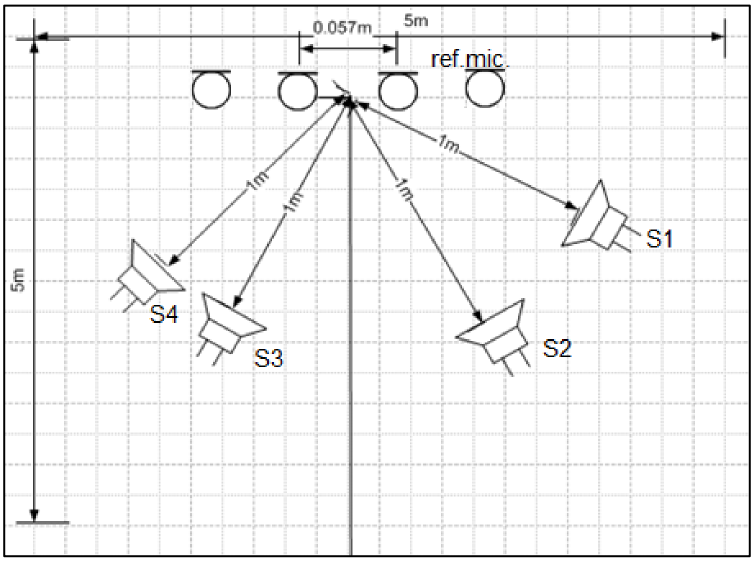


Figure (3.19): 4 MICs, 4 sources environment room

Because using four microphones, the number of nulls for the array factor equals three

$$\boldsymbol{number\ of\ null = number\ of\ MICs. - 1} \qquad (3.1)\ [37]$$

Therefore, this property can use to implement the system in two ways:

i. Uniform Microphone Array Beamforming:-

When used a single tone signals to test the system, figure (3.20) shows the simulation system. In this system four sine wave generators and four calculation blocks are used, each microphone must sense four source signals.

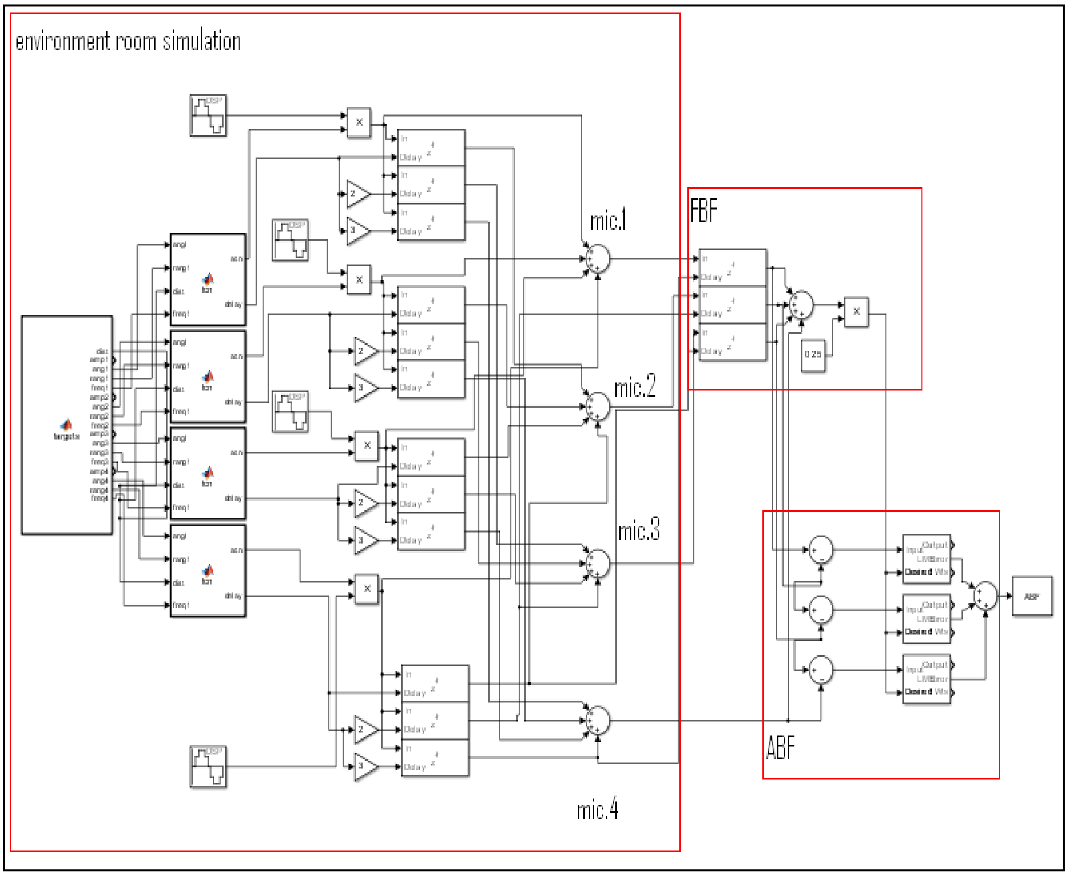


Figure (3.20): Simulation system for 4 MICs, 4 sources uniform phase and amplitude tested by single tone signals

As shown in figure (3.20) four signal generators are used to each of them needs a Matlab function calculation block to calculate the required parameter.

After feed, the parameters definition block with the parameters stated above figure (3.19), the (AF) was calculated by the array factor calculation units as shown in figure (3.21) and the input signals to the four microphones are shown in figure (3.22).

figure (3.21): Th broadside in case phas

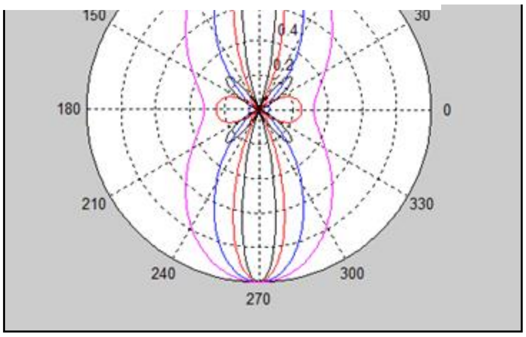
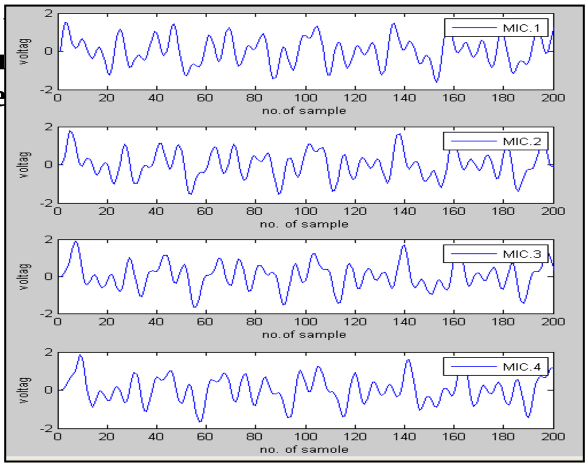


figure (3.22): The input tone signals microphone in case (4 MICs, 4 sou uniform phase and amplitude



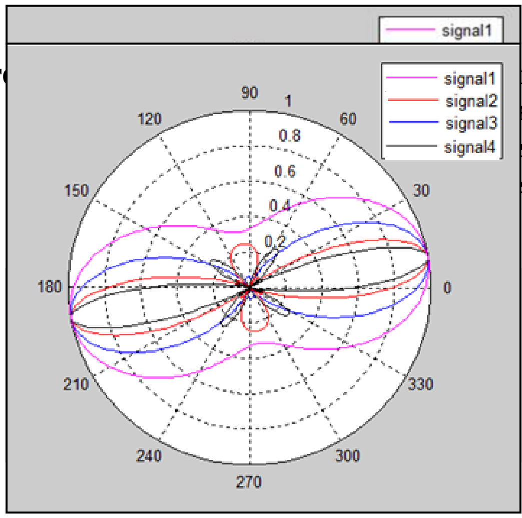
As well as the first microphone assumed as a reference, its input will be the summation of the four signals without delay, while the input of the other microphones will be the summation of the four signals but each signal with delay calculated by the four calculation blocks.

In the (FBF) stage, the output of the reference microphone and the 2nd and 3rd microphones delayed by the desired signal time delay and then added to the output of the second microphone to steer the direction of the (AF) to the desired signal.

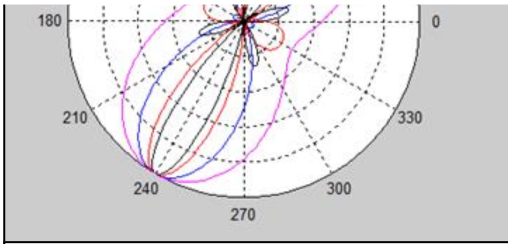
In the (ABF) stage, each microphone output signal will be subtracted from the output of the next microphone, and then one of these outputs will represent one of the adaptive filter inputs. While the output of the (FBF) represents the second input of the adaptive filter, C D

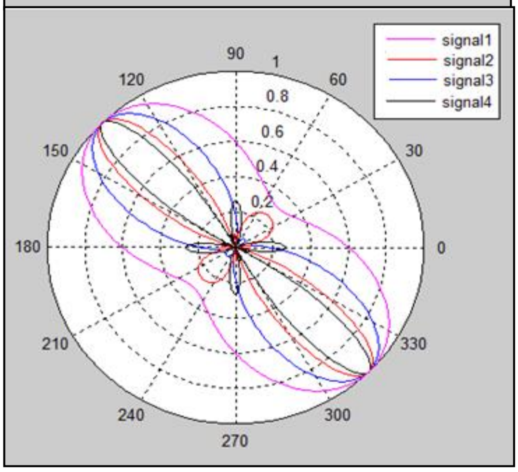
the summation output of the adaptive filters is the output of microphone array beamforming.

Figure (3.23) showing the (AF) for the signal when steering to the desired signal. Figure (3.24) shows the input signals and output signals of the system.



Figur **n case (4 MICs, 4 sources)uniform amplitude** **se array** **B steering at θ2 = 60degree** **D steering at θ4 = 135degree**





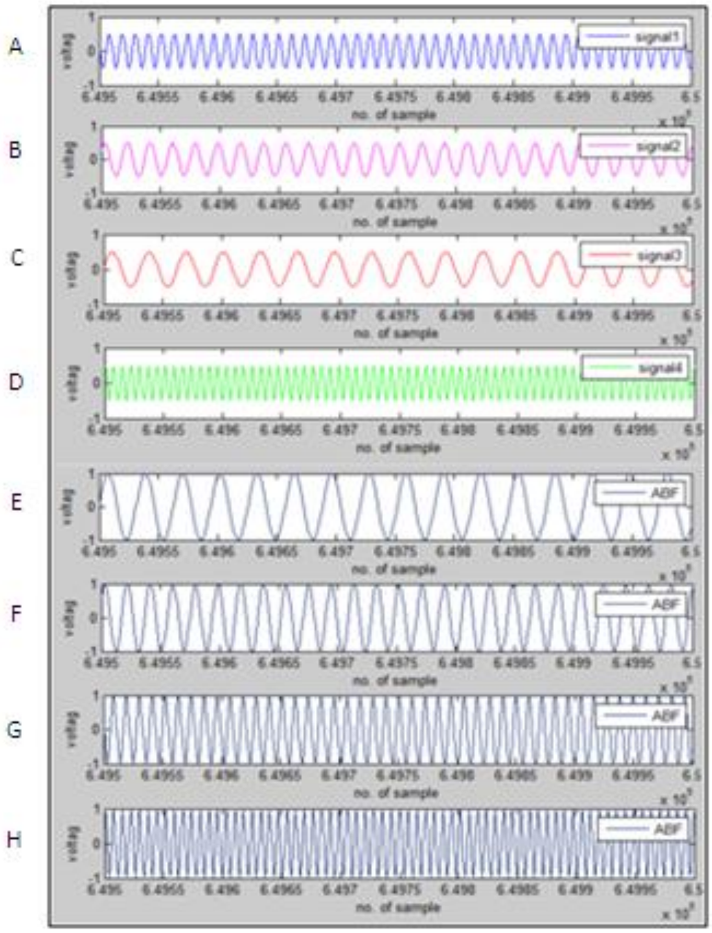


Figure (3.24): (ABF) (single tone inputs signals) & outputs signals in case (4 MICs, 4 sources) (uniform phase and amplitude ABF)
A ton signal1 B ton signal2 C ton signal3 D ton signal4
E (ABF) output when the beam direction set toward the signal at θ_1
F (ABF) output when the beam direction set toward the signal at θ_2
G (ABF) output when the beam direction set toward the signal at θ_3
H (ABF) output when the beam direction set toward the signal at θ_4

When a speech signals are applying in system in figure (3.20), four multimedia file blocks are replaced the four signal generators. Also four digital filters are used as shown in figure (3.25).

The input signal at each microphone is shown in figure (3.26), while the input speech signals and output of the system for each case are shown in figure (3.27).

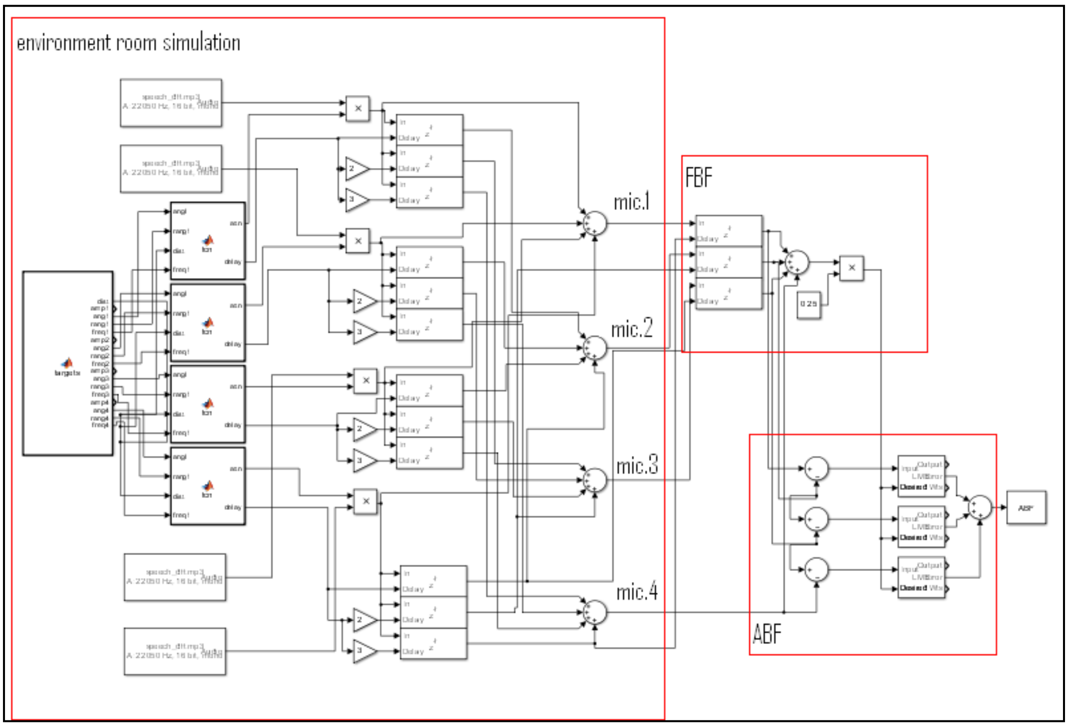


figure (3.25): Simulation system for 4MICs, 4 sources uniform phase and amplitude tested by speech signals

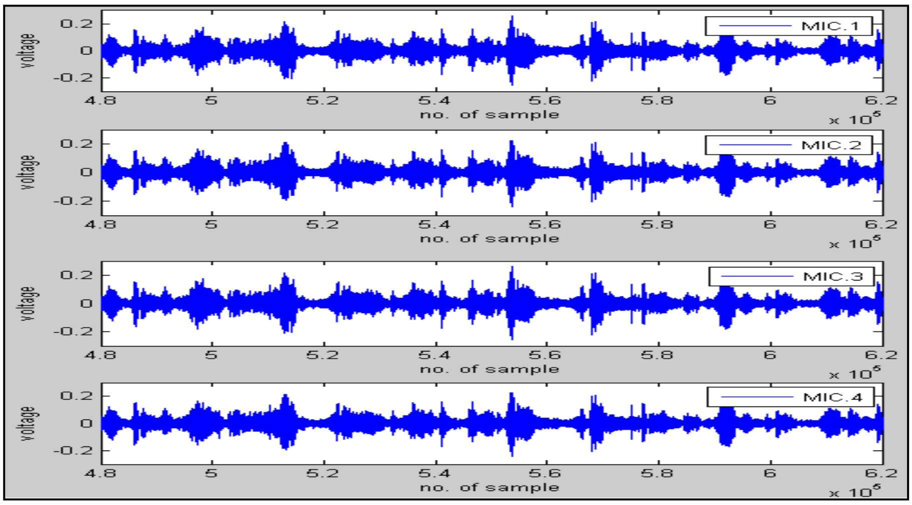


Figure (3.26): The input speech signals to each microphone in case (4 MICs, 4 sources) uniform phase and amplitude

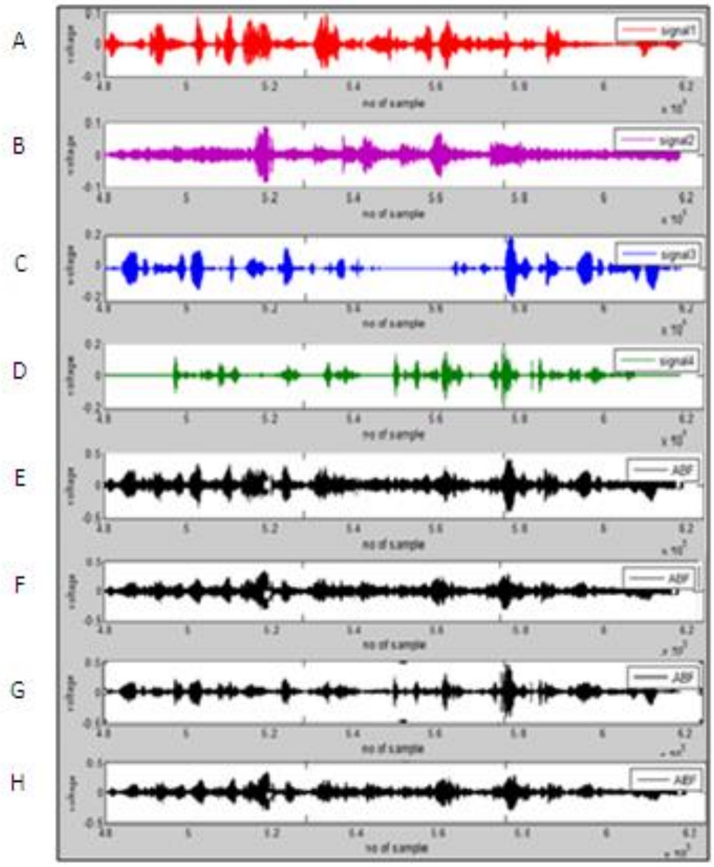


Figure (3.27): (ABF) (speech inputs signals) & outputs signals in case (4 MICs, 4source) (uniform phase and amplitude ABF)
A speech signal1 B speech signal2 C speech signal3 D speech signal4
E (ABF) output when the beam direction set toward the signal at θ_1
F (ABF) output when the beam direction set toward the signal at θ_2
G (ABF) output when the beam direction set toward the signal at θ_3
H (ABF) output when the beam direction set toward the signal at θ_4

ii. Non-uniform Microphone Array Beamforming:-

To enhanced signal to interference ratio, non-uniform amplitude and phase (MABF) are used. The way to construct this system has been done by calculating the parameter of the phase and amplitude by using (LMS) algorithm, this algorithm can build using a Matlab function. Non-uniform amplitude and phase (MABF) system is shown in figure (3.28).

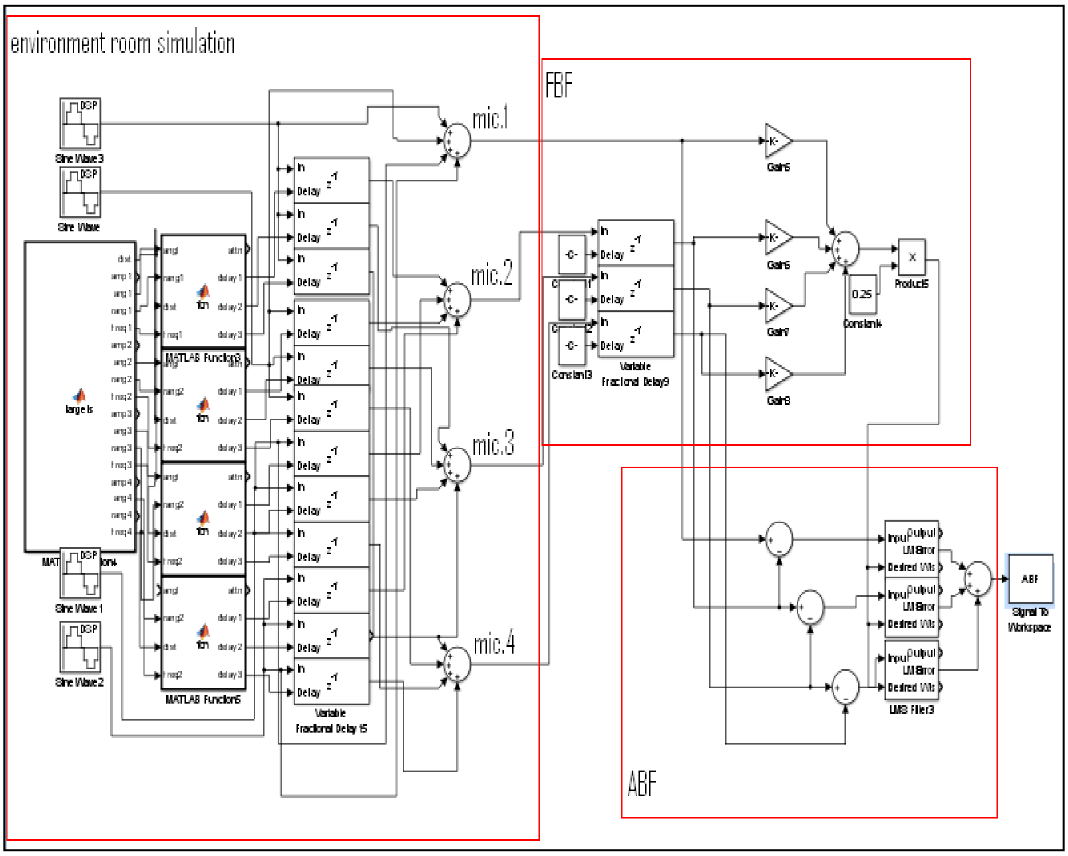
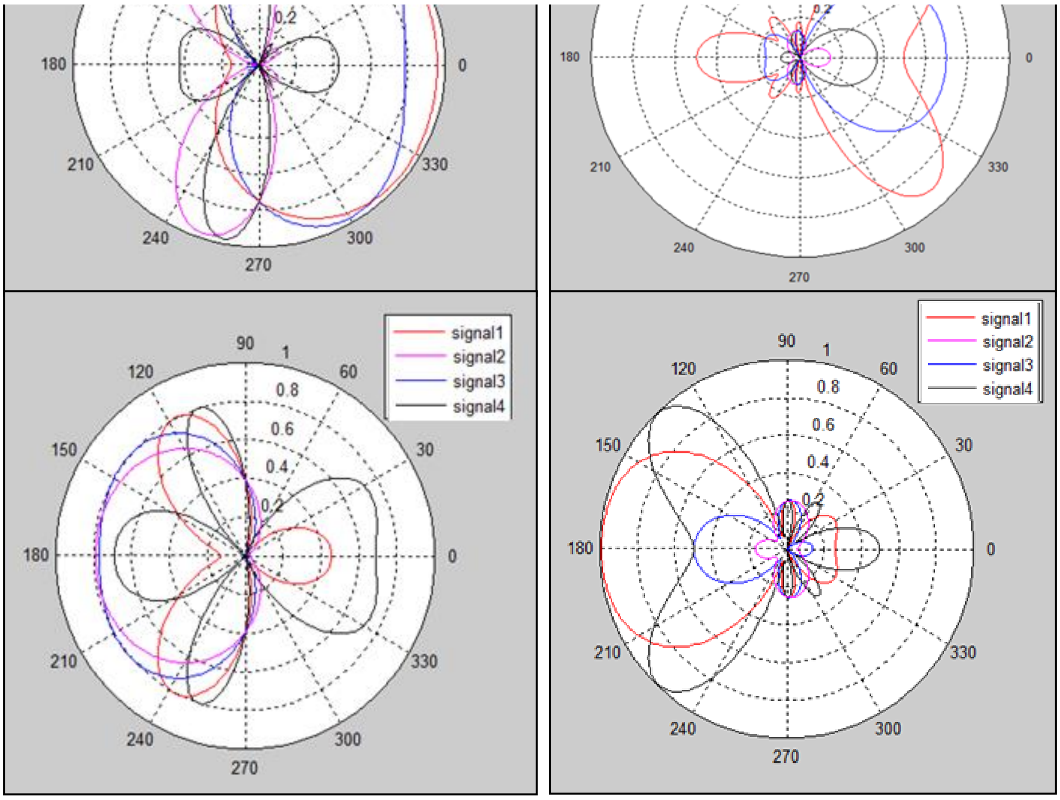


Figure (3.28): Simulation system for 4 MICs, 4 sources non-uniform phase and amplitude tested by single tone signals

As shown in figure (3.28) above the environment room part consists of many units such as a parameter definition block , four blocks to calculate (attenuation, amplitude and phase of each microphone that are calculated by using (LMS) algorithm, and desired signal angle), four single tone signal generators, and the array factor calculation units this calculate after adding amplitude and phase for each microphone.

The (AF) for signals when steering angle (10, 60, 120 and135) degree for the non-uniform amplitude and phase (MABF) system are shown in figure (3.29).

Figure (3.29): The array factor for signals in case (4 MICs, 4 sources) non-uniform phase and amplitude
A steering at θ1 = 10degree B steering at θ2 = 60degree
C steering at θ3 = 120degree D steering at θ4 = 135degree



Now when running the simulation system the output (ABF) shown in figure (3.30).

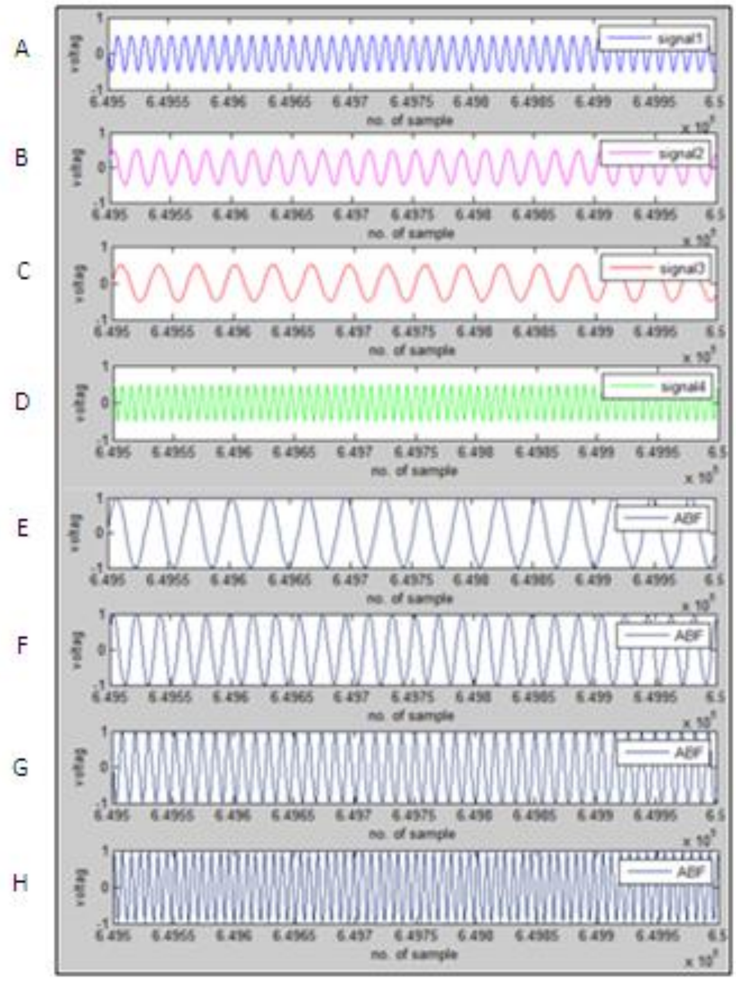


Figure (3.30): (ABF) (single tone inputs signals) & outputs signals in case (4 MICs, 4source) (non uniform phase and amplitude ABF)
A ton signal1 B ton signal2 C ton signal3 D ton signal4
E (ABF) output when the beam direction set toward the signal at θ_1
F (ABF) output when the beam direction set toward the signal at θ_2
G (ABF) output when the beam direction set toward the signal at θ_3
H (ABF) output when the beam direction set toward the signal at θ_4

One can be noticed from figure (3.30) above, that the output for uniform and non-uniform amplitude and phase (MABF) system are not different because used a singleton, but the dissimilar demonstrate when used speech signals that show in figures later.

Figure (3.31) shows the non-uniform amplitude and phase (MABF) system when applying the speech signals in this system.

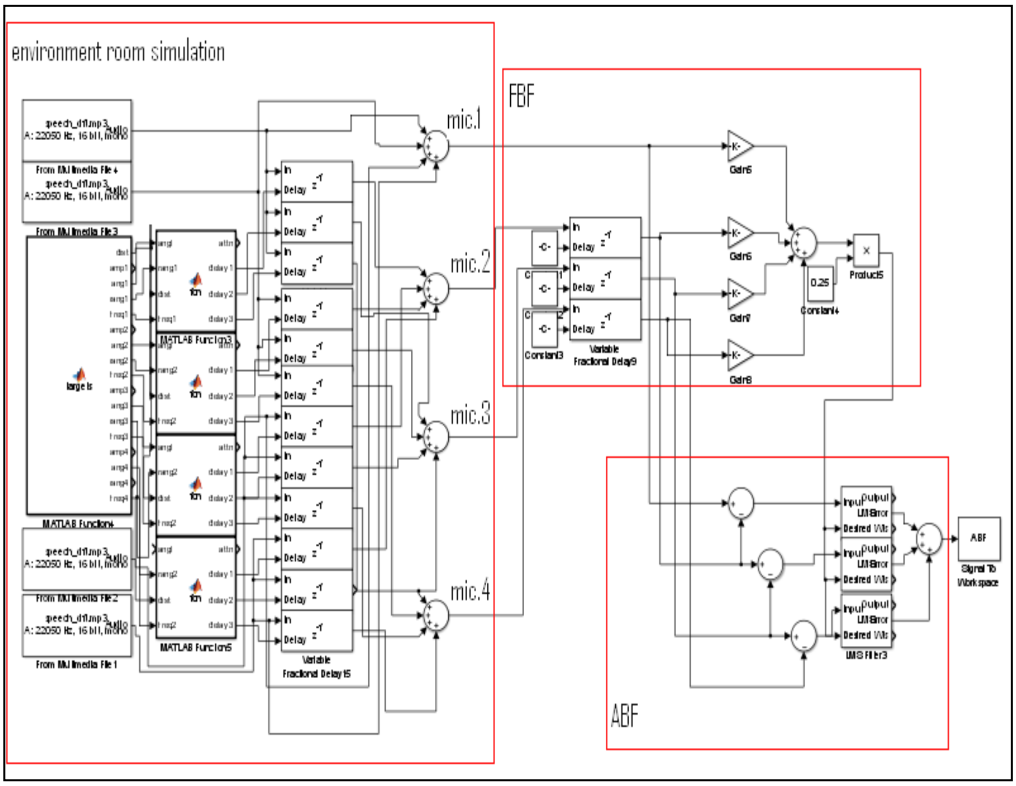


Figure (3.31): Simulation system for 4 MICs, 4 speech sources nonuniform phase and amplitude beamforming system tested by speech signals

Inputs speech signals and outputs for the adaptive non-uniform phase and amplitude (MABF) system are shown in figure (3.32).

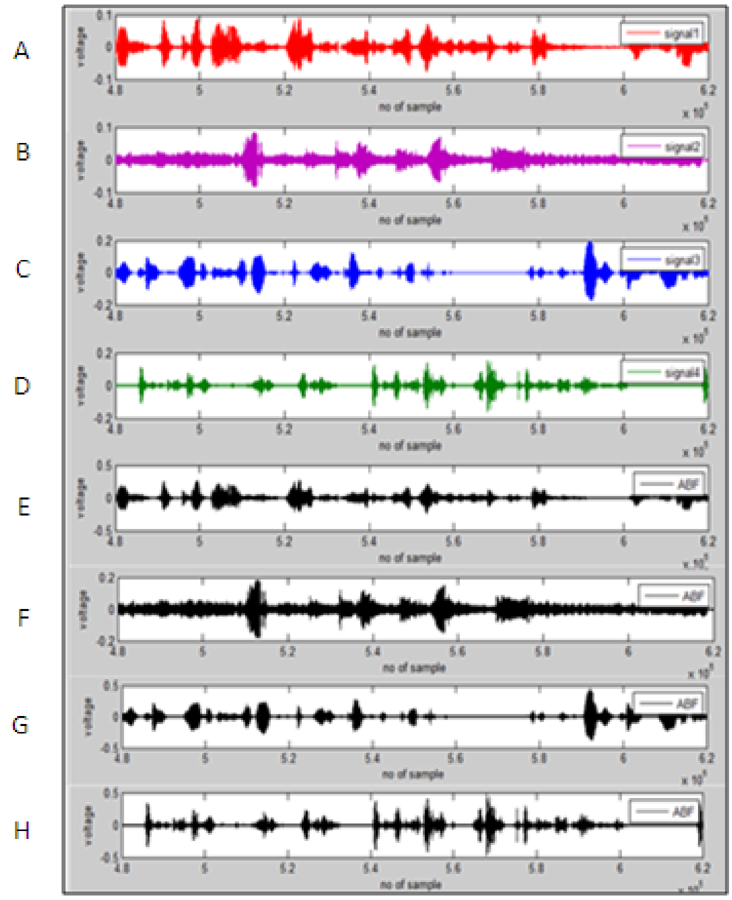


Figure (3.32): (ABF) (speech inputs signals) & outputs signals in case (4 MICs, 4source) (non uniform phase and amplitude ABF)
A speech signal1 B speech signal2 C speech signal3 D speech signal4
E (ABF) output when the beam direction set toward the signal at θ_1
F (ABF) output when the beam direction set toward the signal at θ_2
G (ABF) output when the beam direction set toward the signal at θ_3
H (ABF) output when the beam direction set toward the signal at θ_4

From figure (3.32) above one can be seen that when using non-uniform amplitude and phase (MABF) system, with (LMS) algorithm, the system output contains the desired signal with fewer interference signals.

3.3 Calculation Of The Signal To Noise Ratio For The Simulated Systems:-

The goal for used several ways to design the system is to improve signal to interference ratio.

To calculate the signal to interference ratio for the system, the power signal of output in all stages in (MABF) should be calculated [38].

The steps which used to calculate Δ(S/I) for the first system (uniform two MABF):

1. Calculate the power of each signal at the source for no. of sample equal 220188 the power of

signal$_{desired}$ = 67.687w, signal$_{interference}$ = 57.84w.

$$(S/N)_{\text{at source}} = \frac{67.68}{57.84} = 1.17 = 0.69\text{dB}. \quad (3.2)$$

2. Calculate the power of each signal at the microphone and then calculate S/N.

$$(S/N)_{\text{at MIC}} = \frac{67.68*0.5}{57.84*0.82} = 0.7 = -1.55\text{dB} \quad (3.3)$$

3. Calculate the power of each signal at the output of (DAS) stage.

$$\text{Signal}_{desired} = 67.68 * 1 \text{ , Signal}_{intrference} = 57.84 * 0.5 \quad (3.4)$$

$$(S/N)_{\text{at delay\&sum}} = \frac{67.68*1}{57.84*0.5} = 2.3 = 0.36\text{dB}.$$

signal$_{delay\&sum}$ = desired signal + interference signal (3.5)

= 67.68 + 28.9

= 96.58w = 19.8dBw

4. Calculate the power of interference signals output the (BM) stage for the same no. of sample

signal $_{B.M}$ = 5.62w = 7.49dBw

5. When stable the parameter of adaptive filter calculate the output signal for it.

Signal$_{adaptive\ filter}$ = signal$_{delay\&sum}$ - signal $_{B.M}$ (3.6)

= 96.58 - 5.62

= 89.41w desired and interference signal

$$noise_{adaptive\ filter} = Signal_{adaptive\ filter} - signal_{desired} \quad (3.7)$$

= 89.41 - 67.68

=21.73w

6. Calculate S/N output the (ABF) system.

$$(S/N)_{ABF} = \frac{67.68}{21.73} = 3.11 = 4.63dB.$$

7. Calculate Δ(S/N) of the (MABF) system.

$$\Delta(S/N) = (S/N)_{ABF} - (S/N)_{at\ MIC} \quad (3.8)$$

= 4.63 + 1.55

= 6.18dB

For the second system (uniform four MABF):

When added two microphones for above system the Δ(S/N) equal double or the Δ(S/N) improved 3dB [39], Then

$$\Delta(S/N)_{4\ MIC} = \Delta(S/N)_{2\ MIC} + 3dB \quad (3.9)$$

= 6.18 + 3

= 9.18dB

For the third system (non-uniform four MABF):

This system consists of uniform four (MABF) plus (LMS) algorithm system, the second system adds amplitude and phase to reduce the interference signals at a certain level. Now must calculate the input and output S/N.

$$\Delta(S/N)_{4non\text{-}uniform} = \Delta(S/N)_{4uniforme} + \Delta(S/N)_{LMS} \quad (3.10)$$

$$\Delta(S/N)_{LMS} = output_{LMS\ algorithm} - input_{LMS\ algorithm} \quad (3.11)$$

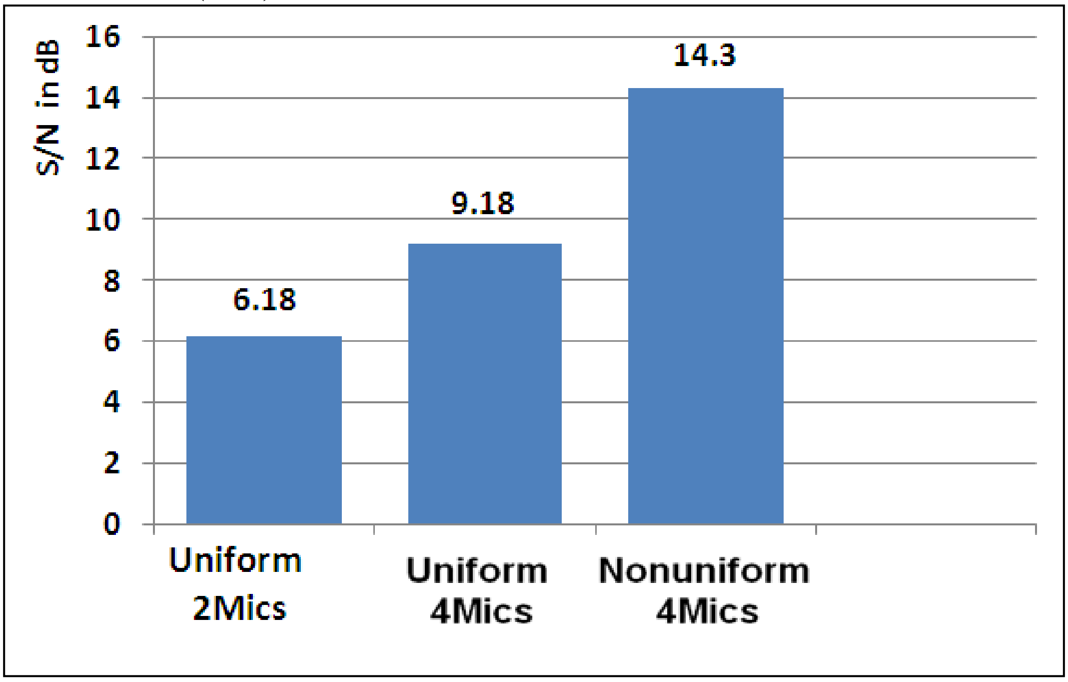
$input_{LMS\ algorithm}$ = -3.75dB

$output_{LMS\ algorithm}$ = 1.37dB

$\Delta(S/N)_{LMS}$ = 1.37 + 3.75

= 5.12dB

$$\Delta(S/N)_{4non\text{-}uniform} = 9.18 + 5.12 = 14.3dB$$



CHAPTER FOUR

HARDWARE IMPLEMENTATION FOR THE MICROPHONE ARRAY BEAMFORMING SYSTEMS USING TMS320C6713 KDS KIT

4.1 Real-Time Digital Signal Processing (DSP):-

DSP processors are most commonly used in real-time signal processing [40]. For real-time processing, external factors must be considered, while when the signal is processed in a non-real-time processing, there are no time limits. For a system that is not real time, data can be recorded on the computer and be using known algorithms used in signal processing. DSP based systems are less susceptible to environmental conditions. DSP Processors have microprocessor characteristics [41]. They are user-friendly, flexible and economical [42]. The perfect reconstruction code has been successfully executed on TMS320C6713 several DSP.

Embedded Integrated Development Environment (IDE) linked to MATLAB and (CCS) program was used to port the (MABF) code on (DSP) kit. (TICCS) objects were used to Transfer Information to and from Code Composer Studio and with the embedded objects [43]. With embedded (IDE) link, we can get information about data and functions stored in (DSP) kit, processor memory, and registers, as well as information about functions in the project.

The (MABF) systems were implemented are using different techniques by many researchers, in this work the TMS320C6713 DSP KIT was used for implementing two types of the (MABF) systems, namely the uniform, and the non-uniform amplitude and phase (MABF) systems.

Each system was implemented by using 2 MICs and 4 MICs, in the case using 2 MICs only one TMS320C6713 DSP KIT was enough, while two TMS320C6713 DSP KIT was used when they are using 4 MICs.

In this chapter, the implementation techniques of these systems are illustrated and these systems results are discussed.

4.2 Uniform Microphone Array Beamforming System Using Two Microphones:-

The implementation of the uniform (MABF) system using two microphones needs TMS320C6713 KDS KIT, three PCs and oscilloscope figure (4.1) and (4.2) shows the block diagram of the implemented system.

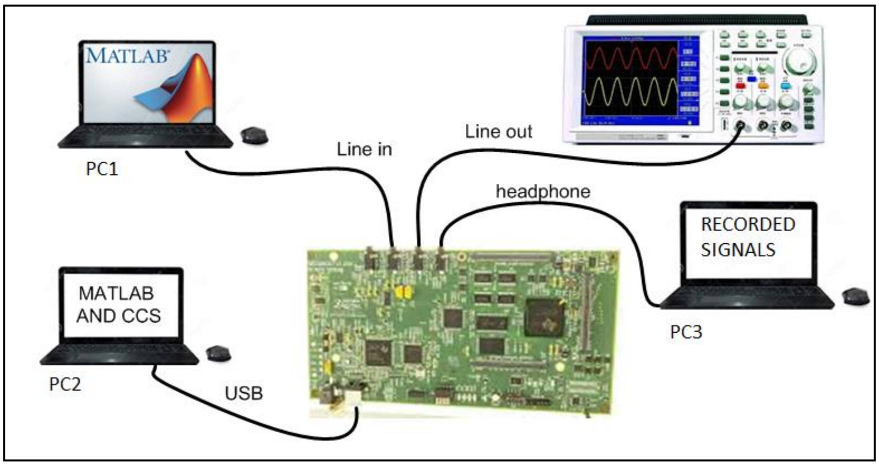


Figure (4.1): Input and output devices connected to TMS320C6713 KDS KIT

Figure (4.2): TMS320C6713 KDS KIT based hardware implementation system of the (MABF) system using two microphones

PC1 is connect to the KDS KIT at stereo line connector is represented at the outputs of the two microphones signals. These signals generated in the PC1 and send to the KDS KIT through the audio outputs of the PC.

Two multimedia file blocks are used to generate the two microphones signals; in each block the signal output of one microphone is recorded. Then used matrix concatenate block synchronization between the two signals, the MATLAB-SIMULINK circuit that used to generate the two microphone signals are shown in figure (4.3).

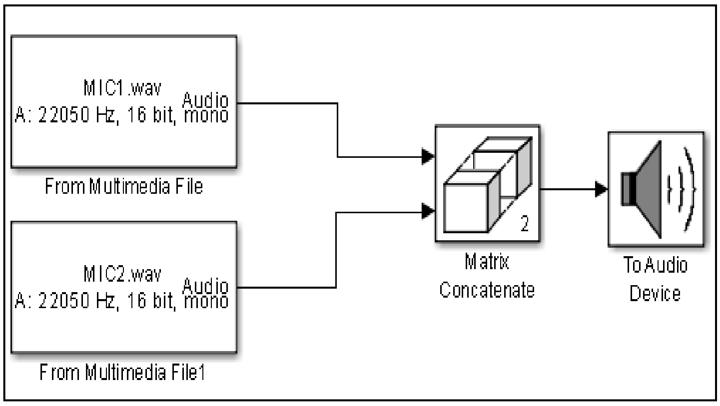
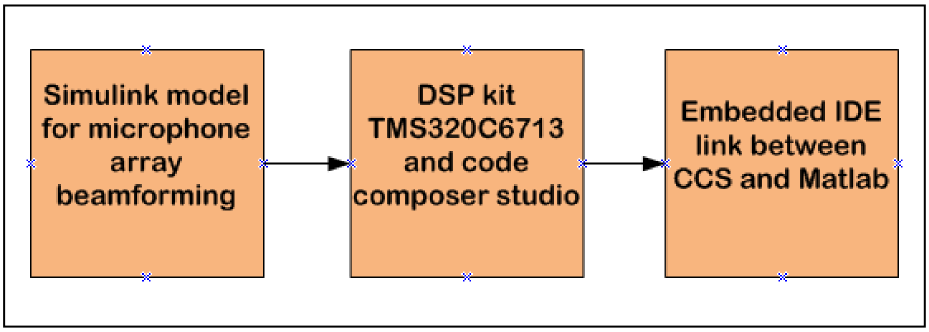


Figure (4.3): MATLAB-SIMULINK circuit that used to generate the two microphone signals (in Pc1)

PC2 represent the host computer for the KDS KIT, in this computer, the KDS KIT codes are generated using MATLAB-SIMULINK program and (CCS) program. This generated code is



loaded to the KDS KIT by USB cable that connects to the PC and the KIT. Figure (4.4) show the steps of implementation system in PC2.

Figure (4.4): Beamforming components for implementation in PC2

The purpose of using PC3 that connect to KDS KIT from headphone connector is to record the output signal for the implemented system.

The oscilloscope is connected to the Line out connector of the KDS KIT to show the output signal.

4.2.1 Uniform Beamforming System Using Two Microphones Two Source:-

In the beginning, two sine wave signals are used as an inputs signals to this system which illustrated earlier in figure (3.2). These inputs signals and the microphones outputs signals which represent the results of the Simulink circuit shown in figure (4.3) (in PC1) are shown in figure (4.5) and (4.6).

Figure (4.5): Two input tone signals
Ch1 (blue line) is tone signal1 with2000Hz
Ch2 (red line) is tone signal2 with1150Hz

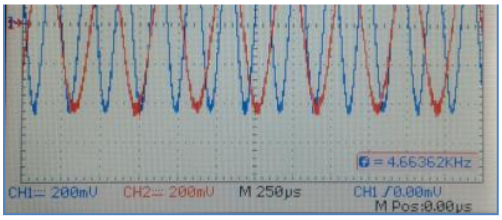


Figure (4.6): The input tone signals to each microphone in case (2 MICs, 2 sources)
Ch1 (blue line) is MIC1 signal (ref.MIC)
Ch2 (red line) is MIC2 signal

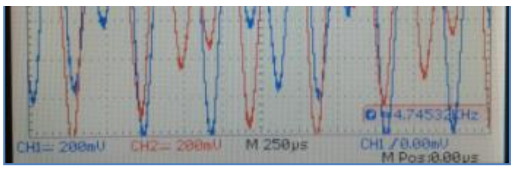


Figure (4.7) shows the MATLAB-SIMULINK circuit which builds for the hardware implementation of the beamforming system with 2 MICs, 2 sources in PC2, while the output signals of this system are shown in figure (4.8).

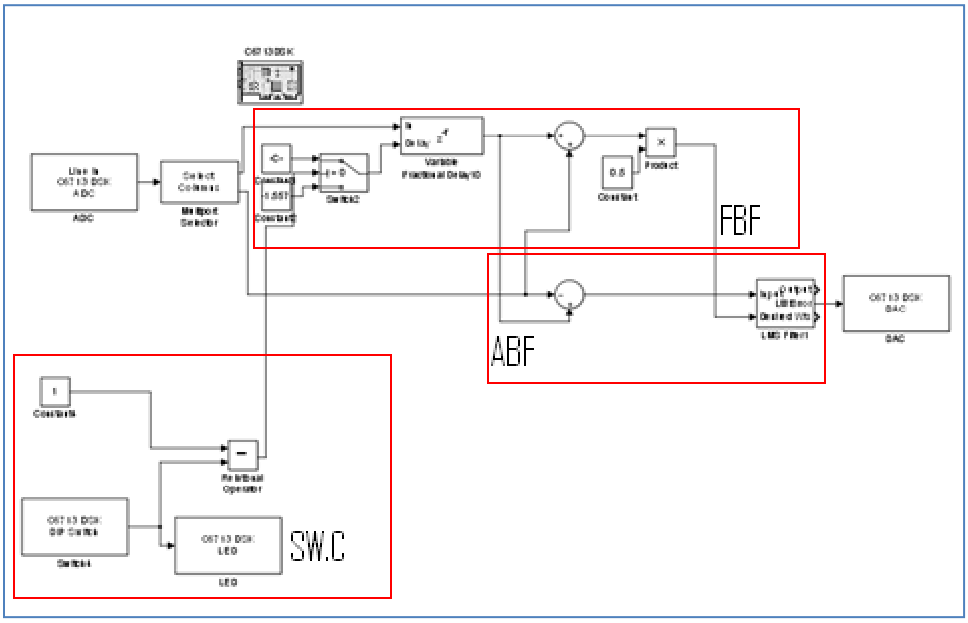
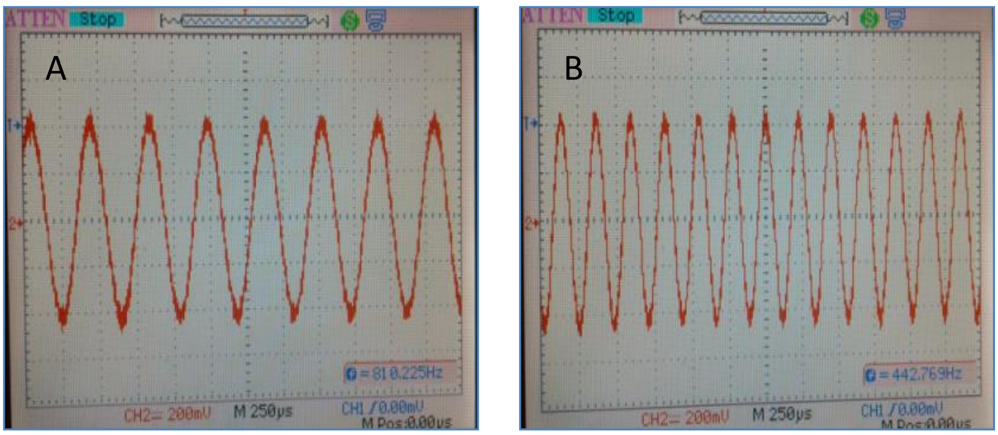


Figure (4.7): Simulation system for 2 MICs, 2 sources using TMS320C6713KDS KIT

In the system above, microphone analog signals are converted to digital signals using Analog To Digital Converter (ADC) block, then the input stereo signal is divided into two mono signals by multiport selector block, these two signals that represent microphone1signal and microphone 2 signal, then these two which are passed to the (FBF) stage and (ABF) stage to get the (MABF) output.

Also, the SW.C representation switches circuit used to converts the time delay to steer the beam to the desired signal by controlling PID switch in the KDS KIT. The Digital To Analog Converter (DAC) block is used to convert the digital output signal of the (MABF) system to analog signal.

After running the system and then tunning the beam between $\theta 1$and then $\theta 2$ the output of the system is shown in oscilloscope display and figure (4.8) show this.



Figuer (4.8): 2 MICs, 2 sources using TMS320C6713KDS KIT output system tested by two tone signals

It is notable that the output signals are identical to the input signal and the reason is that the input signals were single-tone signals.

Now after using two sine wave signals to test the system, two speech signals are used to test the system which illustrated earlier in figure (3.7), these signals are shown in figure (4.9), the output microphones signals of the Simulink circuit shown in figure (4.3) (in PC1) are shown in figure (4.10).

Figure (4.9): Two input speech signals Ch1 (blue line) is tone signal1 with 2000Hz Ch2 (red line) is tone signal2 with 1150Hz

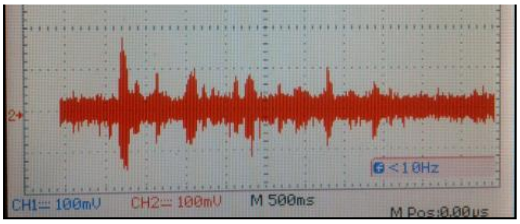
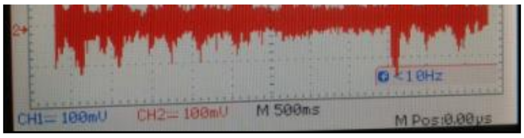
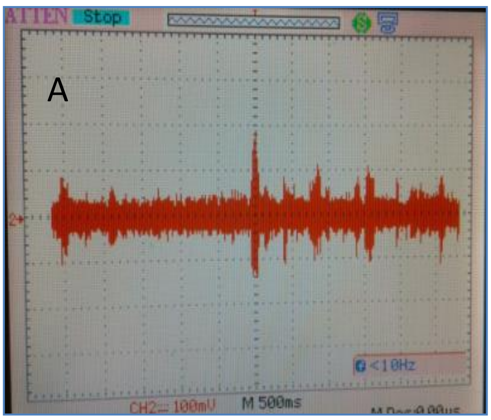
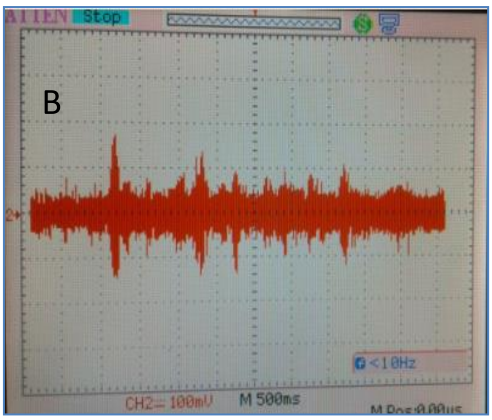


Figure (4.10):The input speech signals to each microphone in case (2 MICs, 2 sources) Ch1 (blue line) is MIC1 signal (ref.MIC) Ch2 (red line) is MIC2 signal



After this signals passed through the system shown in figure (4.6), and then steering the beam to signal1 and then signal2 by control





the system throws the PID switch, the output signals are shown in figure (4.11).

Figuer (4.11): 2 MICs, 2 sources using TMS320C6713KDS KIT output system tested by two speech signals
A (ABF) output when the beam direction the signal at θ_1
B (ABF) output when the beam direction the signal at θ_2

Figure (4.11) show that the output speech signals system are mixed between the desired signal and interference signal that means the system cannot delete all interference signal.

4.2.2 Uniform Beamforming System Using Two Microphones Four Source:-

In the beginning, four sine wave signals are used as an inputs signals to the system shown in figure (3.11). These inputs signals and the microphones outputs signals which represent the results of the Simulink circuit shown in figure (4.3) (in PC1) are shown in figure (4.12), and (4.13).

Figure (4.12): Four input tone signals

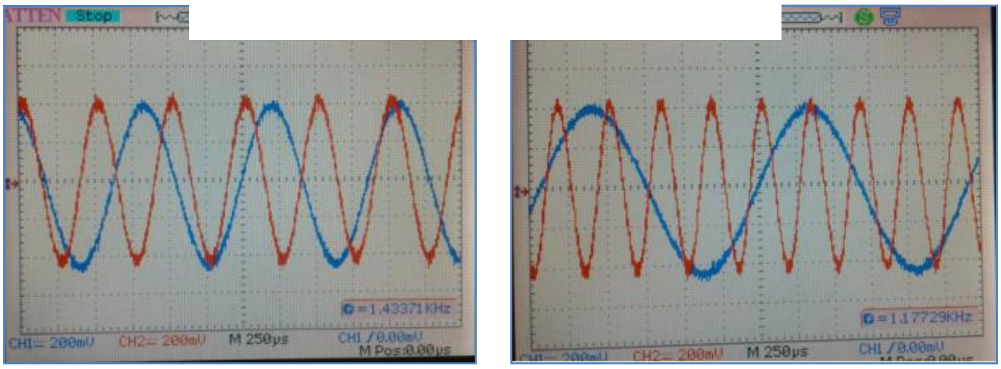


Figure (4.12) shown the input tone signals with frequency 2000, 1150, 3000 and 700 Hz are placed at θ_1 =10degree, θ_2 = 60degree, θ3 = 120degree and θ4=135degree.

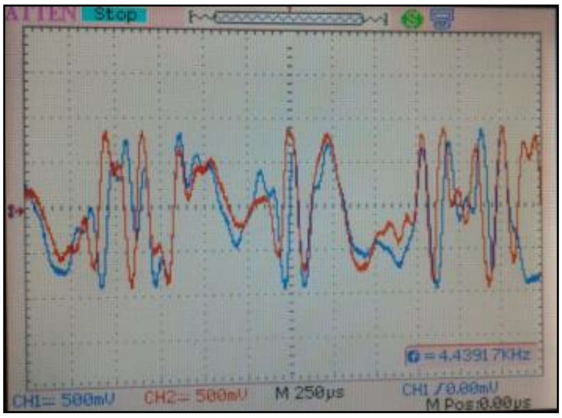


Figure (4.13):The input tone signals to each microphone in case (2 MICs, 4 sources)

Ch1 (blue line) is MIC1 signal (ref.MIC)

Ch2 (red line) is MIC2 signal

builds for the hardware implementation of the beamforming system with 4 MICs, 2 sources in PC2.

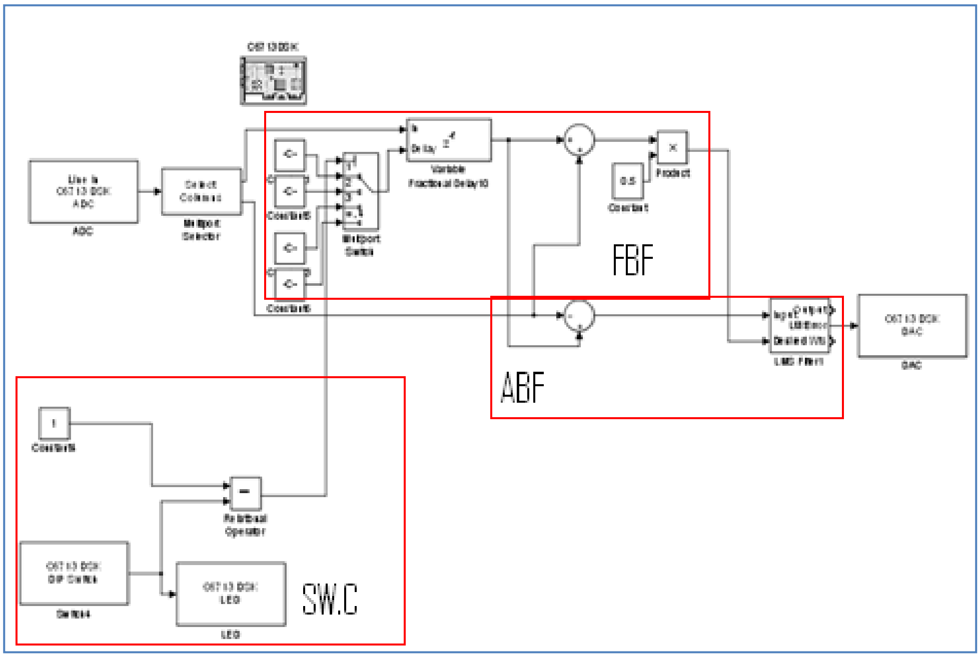


Figure (4.14): Simulation system for 2 MICs, 4 sources using TMS320C6713KDS KIT

It is notable in figure (4.13) that four constant blocks are used to add the delay to steer the beam towered the desired signals by control for SW.C.

Now when control the system throws the PID switch to steer the beam to signal1, signal2, signal3, and signal4. The outputs of the beamforming system of these cases are shown in figure (4.15).

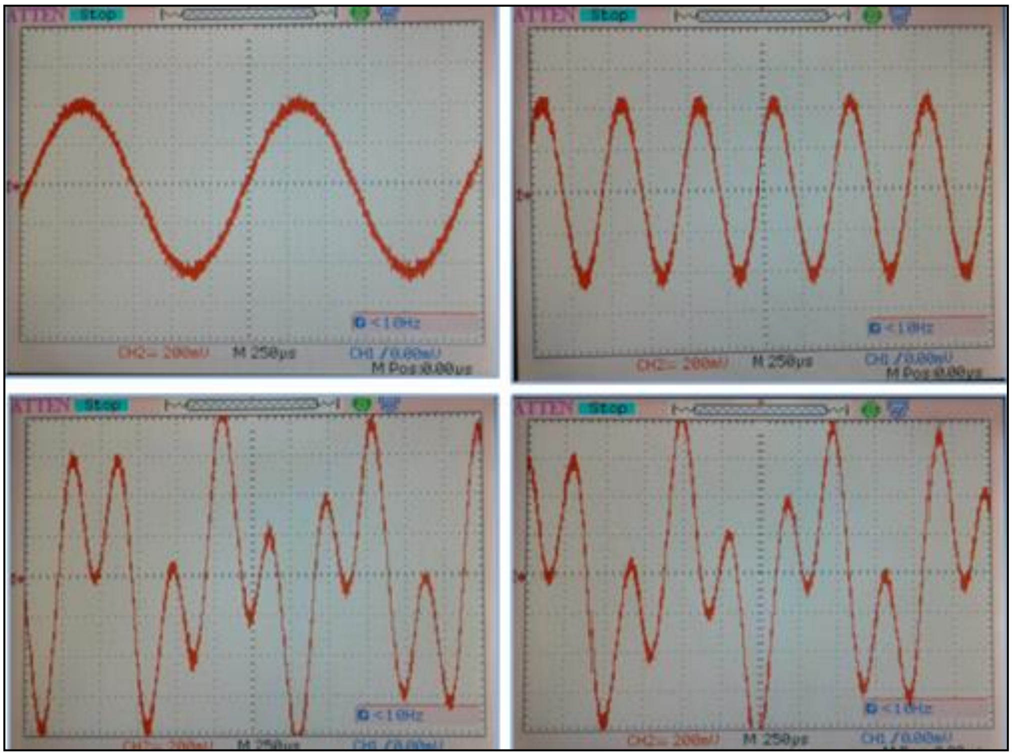
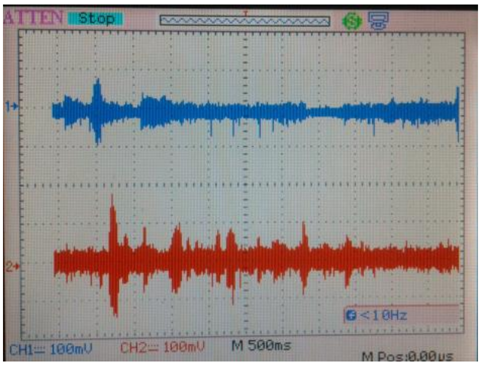
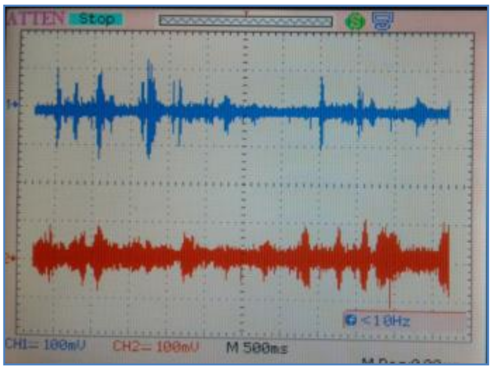


Figure (4.15): 2 MICs, 4 sources using TMS320C6713KDS KIT output system tested by four tone signals
A (ABF) output when the beam direction the signal at θ_1
B (ABF) output when the beam direction the signal at θ_2
C (ABF) output when the beam direction the signal at θ_3
D (ABF) output when the beam direction the signal at θ_4

From figure (4.15) when used tone signals with frequency equal 700, 2000Hz the system canceled all interference signals and when used frequency equal 1150, 3000Hz the output system still contain interference signals.

When four speech signals are used to test the system shown in figure (3.16) and this signals are shown in figure (4.16). The output microphones signals of the circuit shown in figure (4.3) (in PC1) are shown in figure (4.17).





s

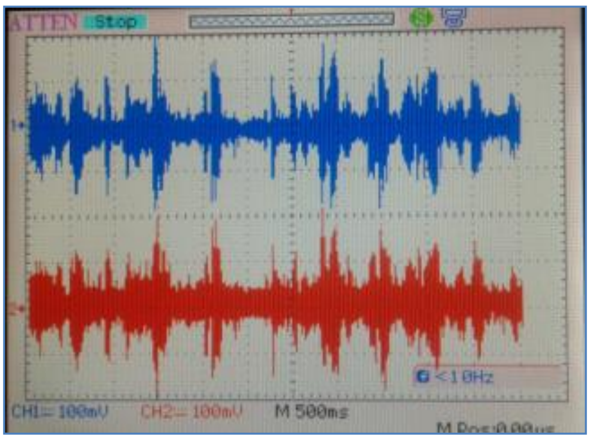


Figure (4.17): The input speech signals to each microphone in case (2 MICs, 4 sources)
Ch1 (blue line) is MIC1 signal (ref.MIC)
Ch2 (red line) is MIC2 signal

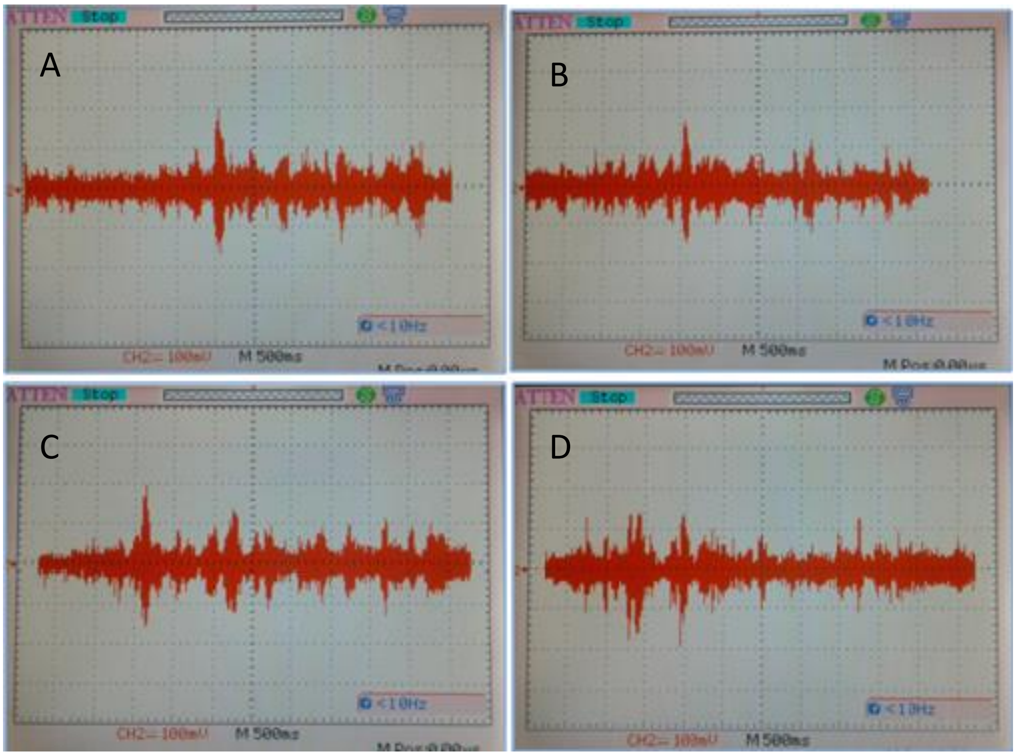
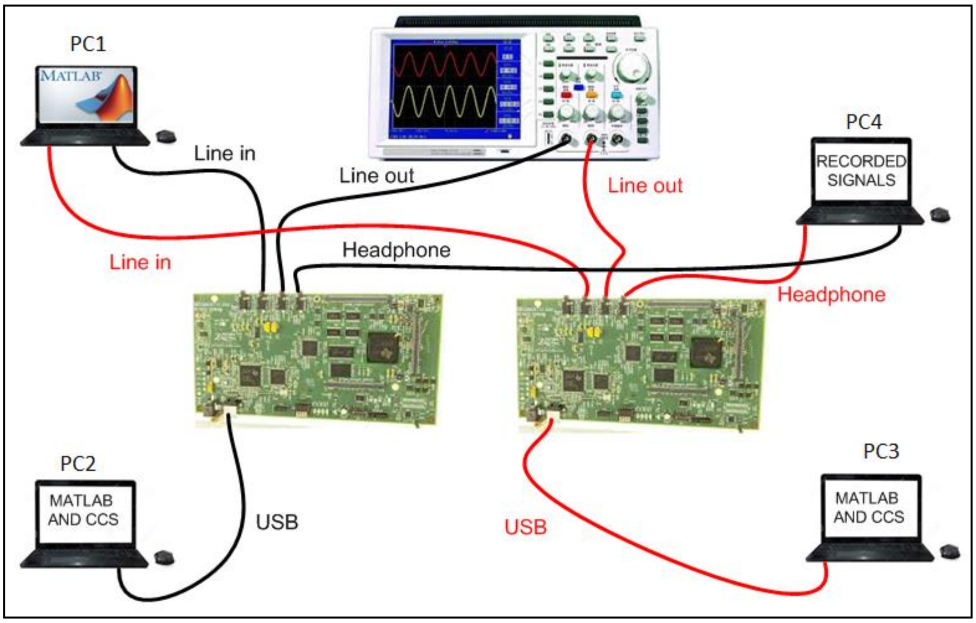


Figure (4.18): 2 MICs, 4 sources using TMS320c6713KDS KIT output system tested by four speech signals
A (ABF) output when the beam direction the signal at θ_1
B (ABF) output when the beam direction the signal at θ_2
C (ABF) output when the beam direction the signal at θ_3
D (ABF) output when the beam direction the signal at θ_4

Figure (4.18) shows that the output of system when to control the beam through the SW.C and change the time delay, This output is the desired signal in addition to the remaining interference that has not been deleted from the system.

4.3 Microphone Array Beamforming System Using Four Microphones:-

In order to implement this system two of the TMS320C6713 KDS KITs are needed, four PCs and oscilloscope. Figures (4.19) and (4.20) show the block diagram of the hardware-implemented beamforming system that uses 4 microphones.



TMS320C6713 KDS

Figure (4.20): Two TMS320C6713 KDS KITs based hardware implementation system of the (MABF) system using four microphones

PC2 which connected to the KDS KIT 1 at stereo line connector represents the outputs of the two microphones signals. These signals generated in the PC1 and send to the KDS KIT1 through the audio outputs of the PC1. Also, PC3 are connected to the KDS KIT 2 at stereo line connector are represent the outputs of the other two microphones signals. These signals generated in the PC1 and send to the KDS KIT2 through the USB audio device connector.

Four multimedia file blocks are used to generate the four microphones signals, in each block the signal output of one microphone is recorded. Then used two matrix concatenate blocks to synchronization between the four signals, each two input signals connected to one matrix concatenate block, the MATLAB-SIMULINK circuit that used to generate the two microphone signals are shown in figure (4.21).

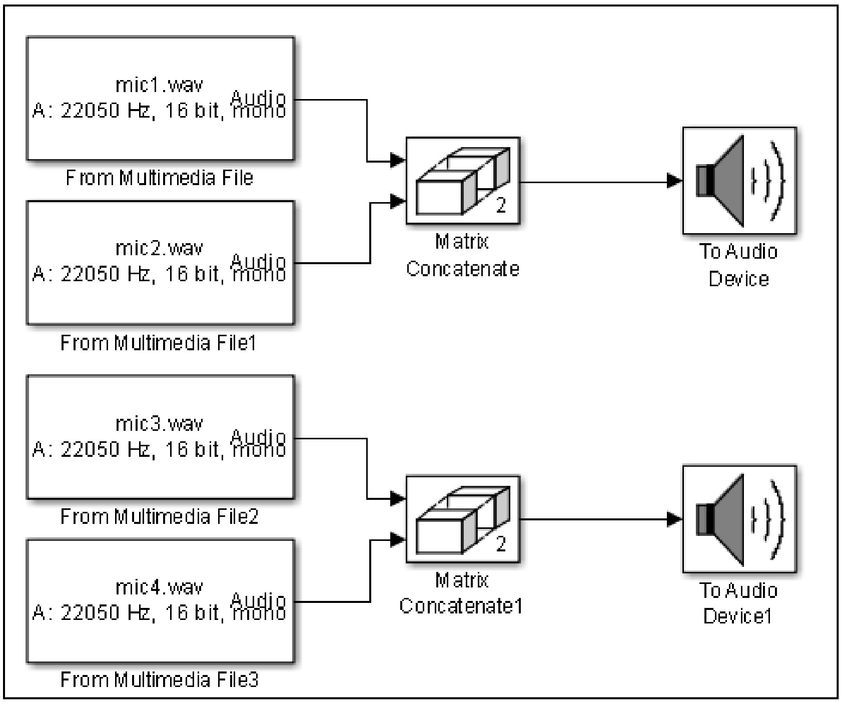


Figure (4.21): MATLAB-SIMULINK circuit that used to generate the four microphone signals (in PC1)

The purpose of using PC4 that connect to KDS KIT from headphone connector is to record the output signal for the implemented system.

PC2 represent the host computer for the KDS KIT1, in this computer, the KDS KIT1 codes are generated using MATLAB-

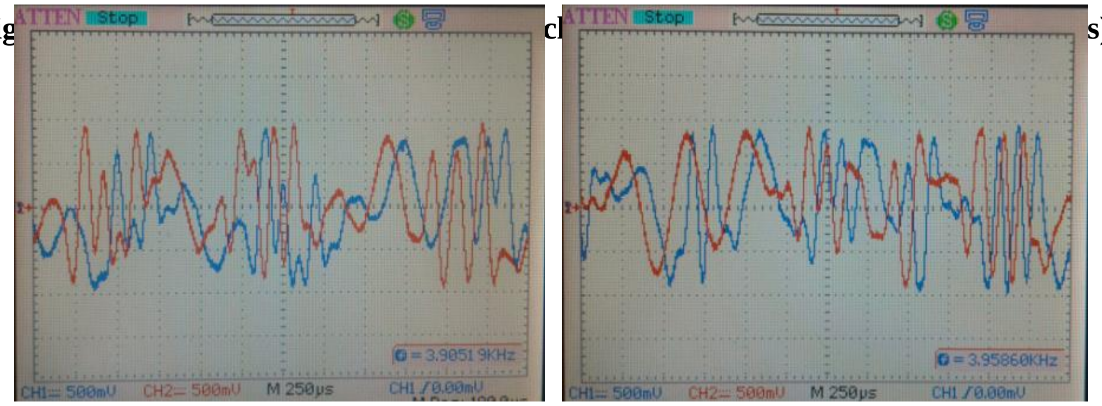
SIMULINK program and (CCS) program. This generated code is loaded to the KDS KIT1 by USB cable that connects to the PC and the KIT1. While PC3 represents the host computer for the KDS KIT2, in this computer the KDS KIT2 codes are generated using MATLAB-SIMULINK program and (CCS) program. This generated code is loaded to the KDS KIT2 by USB cable that connects to the PC and the KIT2.

Line out of KIT1 connector connect to an oscilloscope at channel1 while Line out of KIT2 connector connects to an oscilloscope at channel2 to show the signals by using adding mode.

4.3.1 Uniform Amplitude And Phase Microphones Using Four Input Signals:-

The system shown in figure (3.17) is used in this step, but with some change to the structure that needed. The first change is the conversion of the subtractor blocks in (ABF) path to adaptive filter blocks variable blocking matrix with (CCAFs), then divided the system into two subparts the 1st part apply by the PC2 device to generate the codes for KIT1, the 2nd applies to PC3 to generate the codes for KIT2.

Now when four sine wave signals are used in the system shown in figure (4.12), the input microphones signals apply at Simulink circuit shown in figure (4.21) are shown in figure (4.22).



The two MATLAB-SIMULINK circuits which represent the two part of the hardware system in PC2 and PC3 are shown in figure (4.23) and (4.24). The output signals of the system are shown figure (4.25).

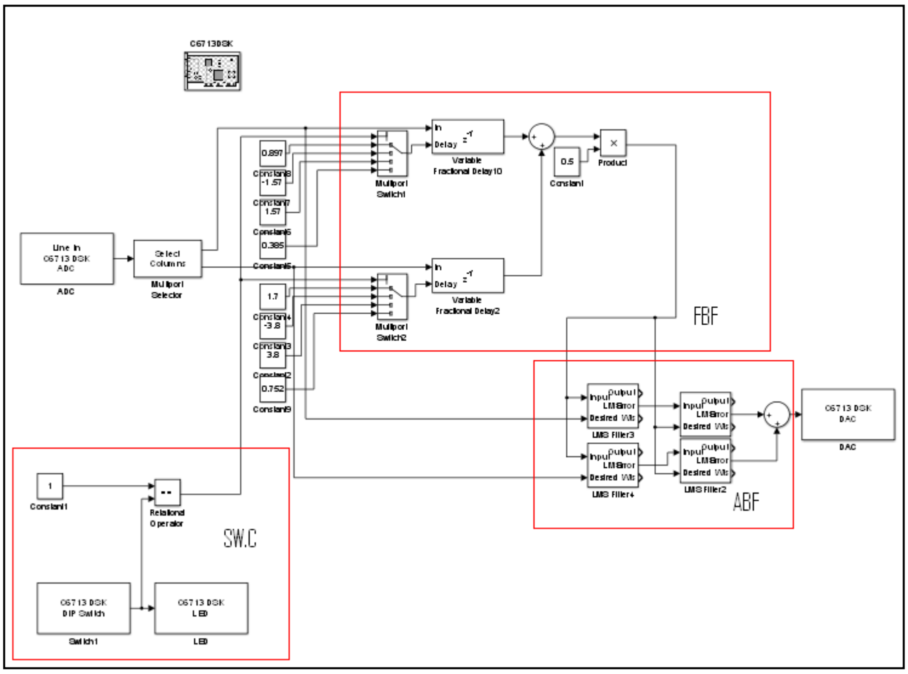


Figure (4.23): Part1 Simulation system for 2 MICs, 4 sources using TMS320C6713KDS KIT1 (PC2)

C

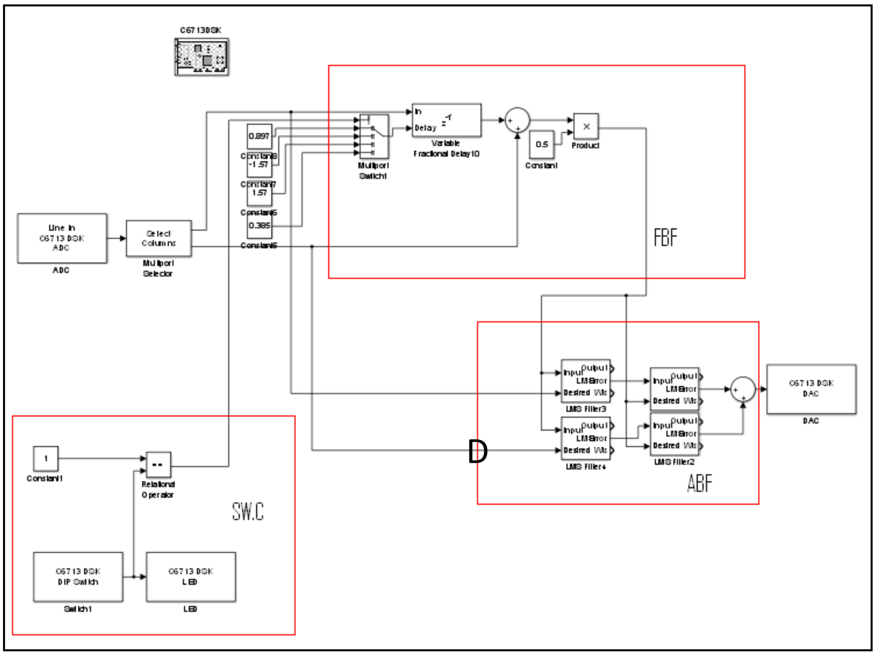
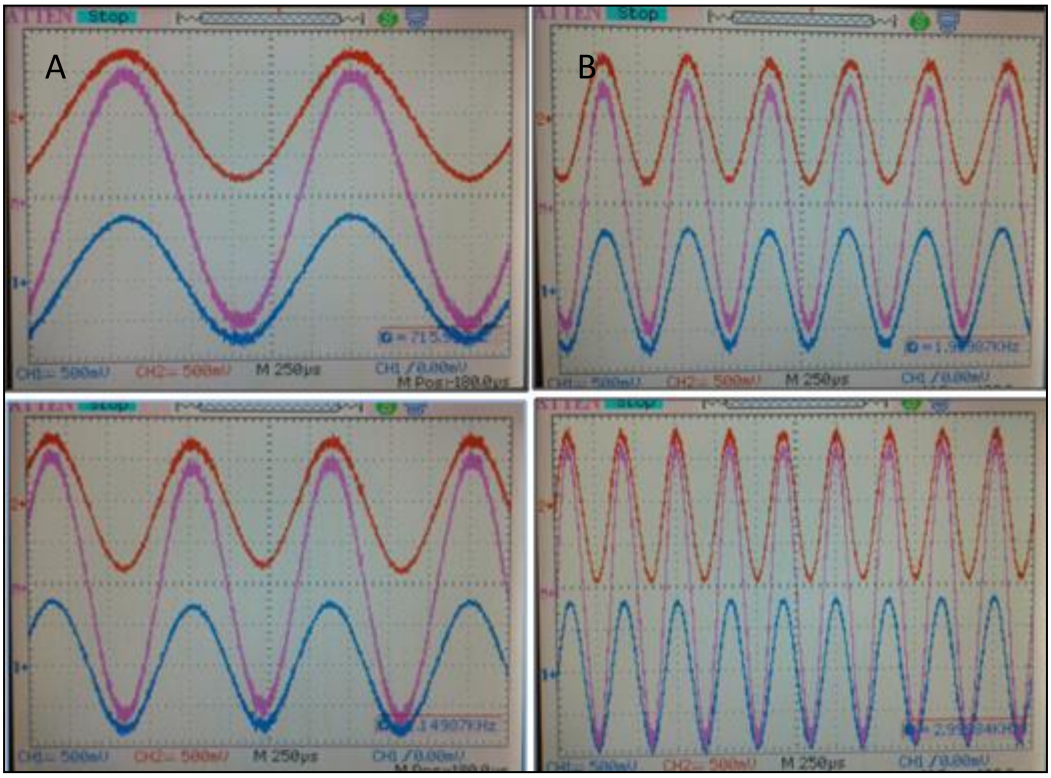


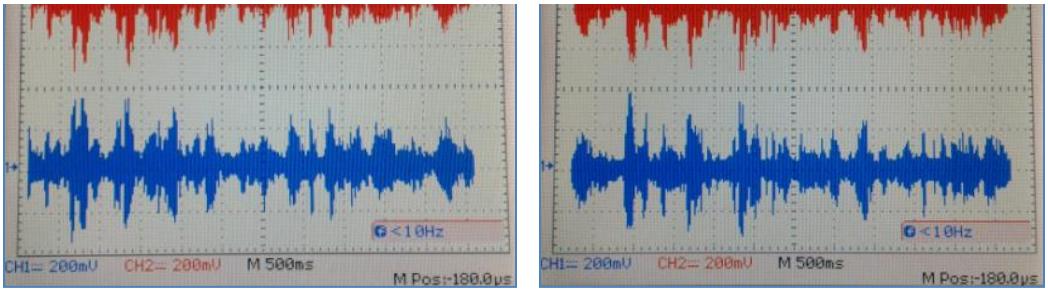
Figure (4.24): Part2 Simulation system for 2 MICs, 4 sources using TMS320C6713KDS KIT2 (PC3)



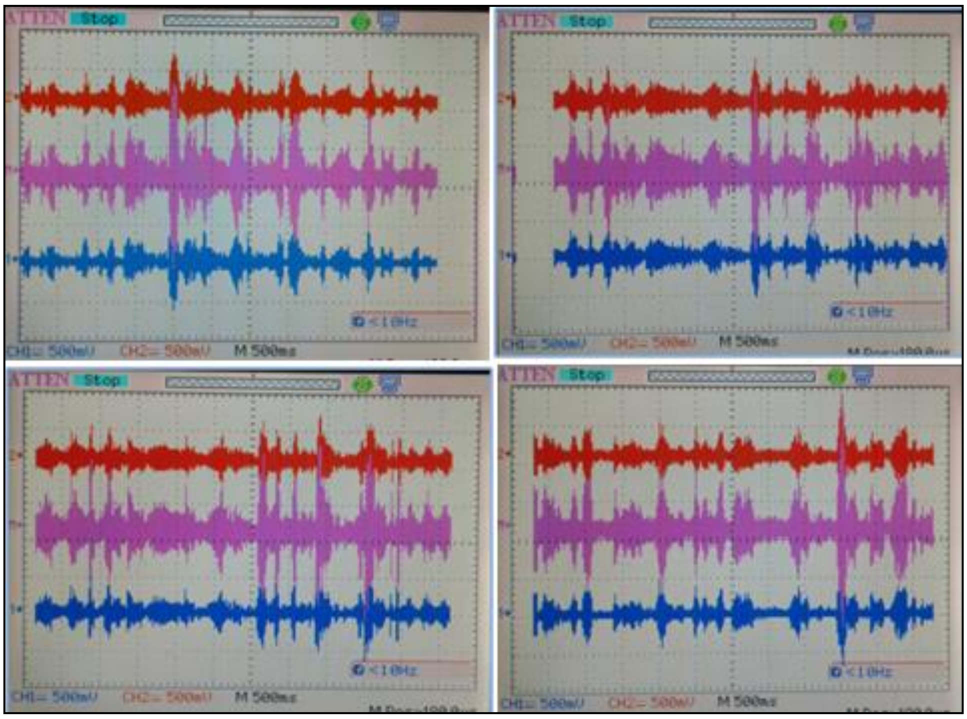
From figure (4.25) when divided the original system and then summation the output of the two subparts, these outputs are same frequency of the desired input signals when the beam steering to its.

Now when using four speech signals instead of the four sine wave signals, the system used is shown in figure (3.25) and this signals are shown in figure (4.16), while the output microphones signals of the Simulink circuit shown in figure (4.21) are shown in figure (4.26).

Figure (4.26): The input speech signals to each microphone in case (4 MICs , 4 sources)



After running MATLAB-SIMULINK circuit which showed in figure (4.23) and (4.24). Are contributing to generating the output signals that shown figure (4.27).

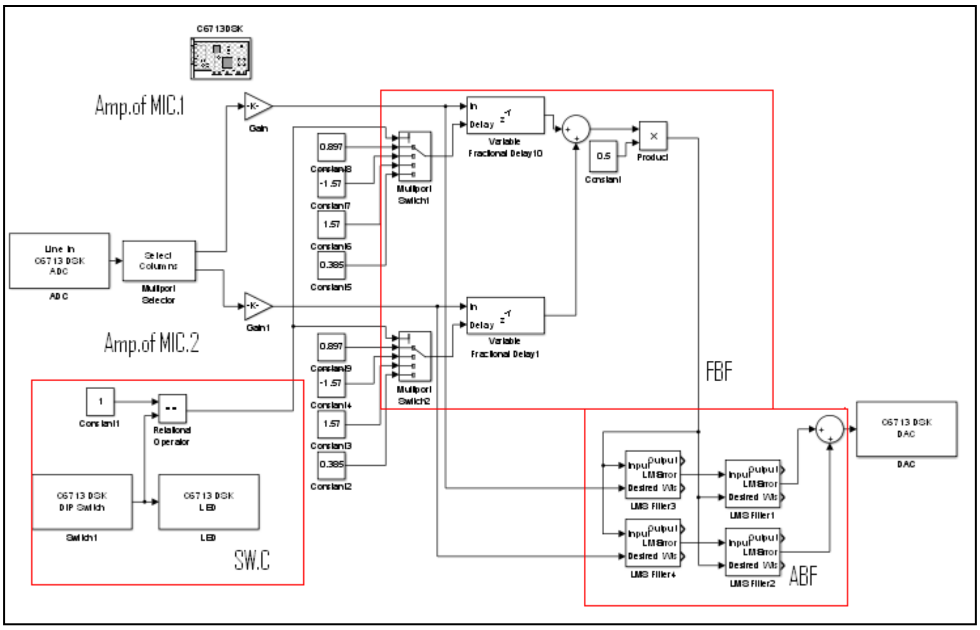


320C6713KDS
ch signals
he signal at θ_1
he signal at θ_2
he signal at θ_3
he signal at θ_4

From figure (4.27) one can be seen that the output of the system (math channel), this output represents the desired signal and the remaining of three interference signals.

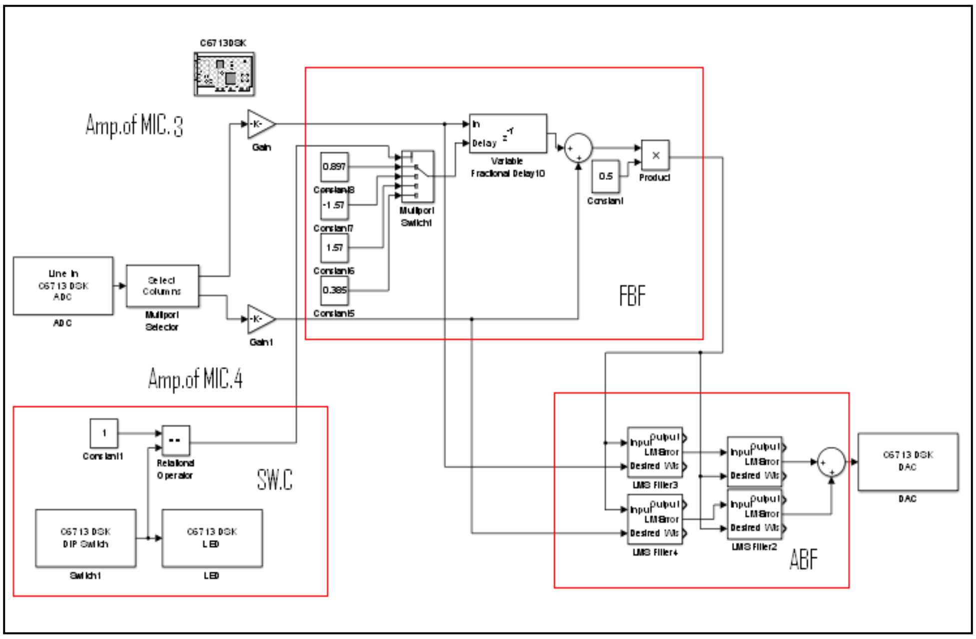
4.3.2 Non-uniform Amplitude And Phase Microphones Using Four Input Signals:-

In order to implement this system, the simulation circuit is shown in figure (4.28) used to generate the KIT1 code in PC2, while the



simulation circuit shown in figure (4.29) used to generate the KIT2 code in PC3.

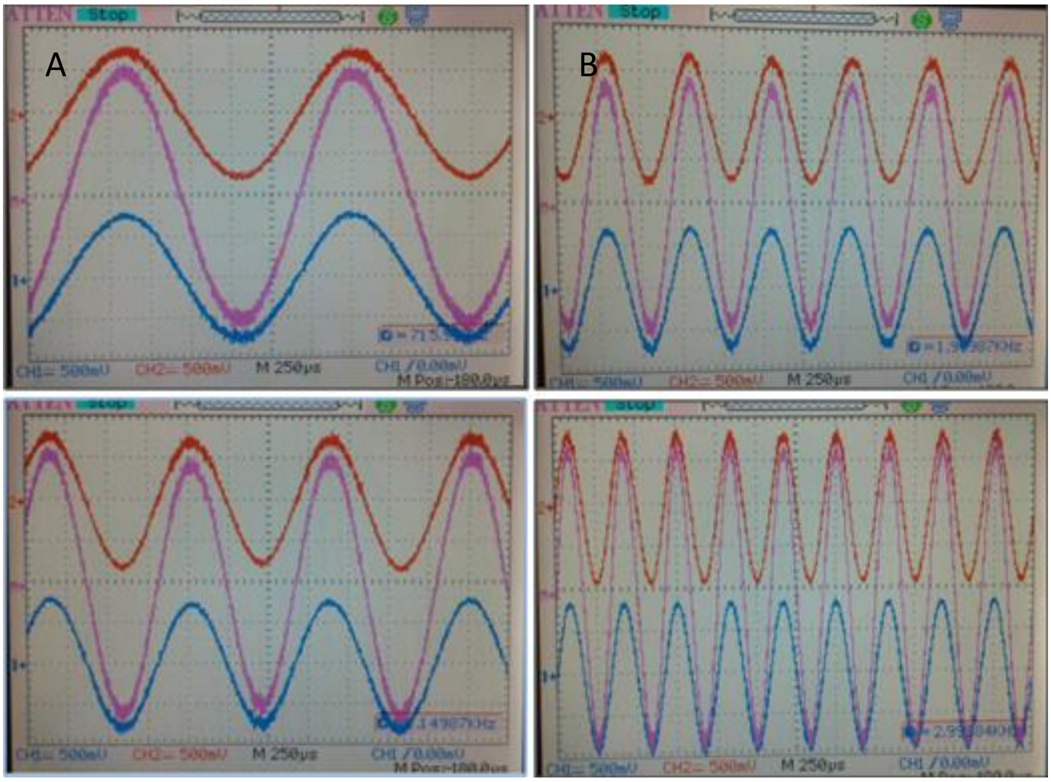
ICs, 4 sources using



C D

MICs, 4 sources using

To test the system by using four sin wave signals and it was shown in figure (4.11) is used, the output microphones signals apply at Simulink circuit shown in figure (4.19) and this signals shown in figure (4.28).

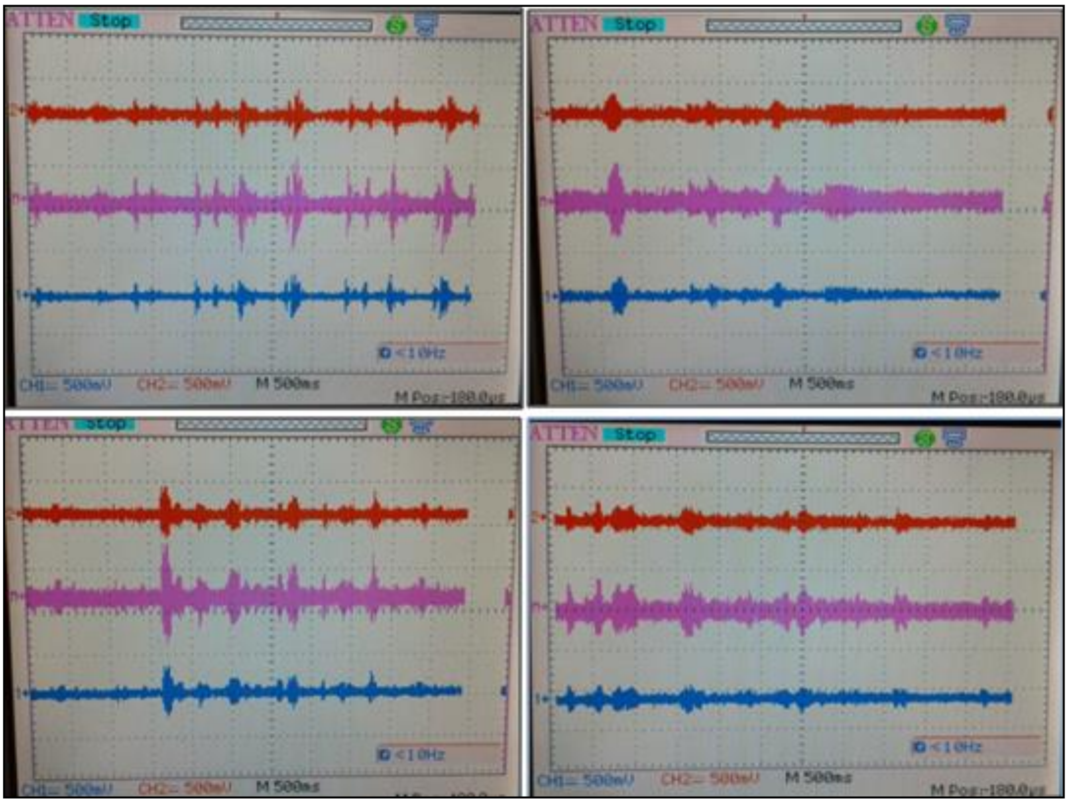


From figure (4.30) one can be note that the output of the system of the uniform array is not different from the non-uniform amplitude and phase (MABF) because used single tone signals.

C D

Now when using four speech signals from the system shown in figure (3.28) are used and this signals shown in figure (4.16), the input microphones signals apply at Simulink circuit shown in figure (4.21) and this signals are shown in figure (4.26). The output of each KIT and the output system is shown in figure (4.31).

A B



For speech signals and when implementation non-uniform amplitude and phase the interference signals in output system are fewer Compared to the output of another system.

When recoded the output signals for each system and synchronization with input signals subtracted between their power signals to get the power noise signal at output system to calculate the signal to noise ratio for each system.

In practical the signal to noise ratio for system, it showed in figure (4.32)

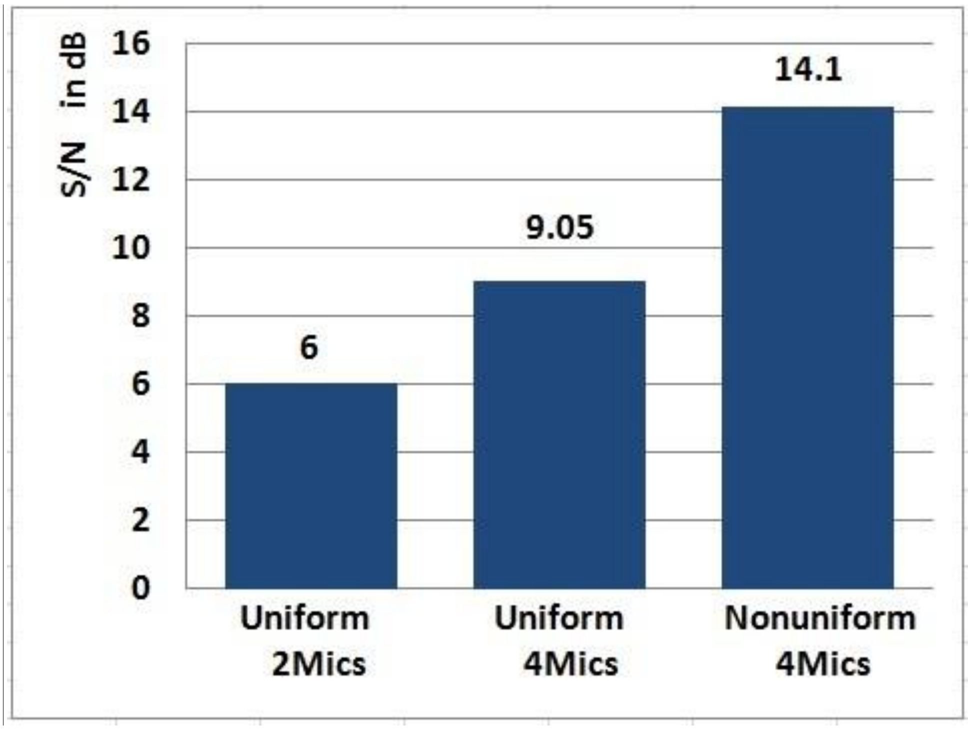


Figure (4.32): The (S/N) for the practical systems

CHAPTER FIVE

CONCLUSIONS AND FUTURE WORK

5.1 Conclusions:-

From the previous work it can be concluded the following:-

1. In general, the (MABF) system is an efficient technique to cancel the unwanted interference signals from input signals, this process done by steering the null of the beamforming array toward the unwanted signals.
2. MATLAB-SIMULINK program is so suitable and reliable for simulation many electrical and electronic systems, one of them is the microphone array beamforming system.
3. The results of the simulation systems when tested by tone signals are so ideal and accurate with the theory of this technique. While the results of this system when tested using speech signals are not ideal as well as the results of these systems when tested with tone signals.

4. The calculation of the S\ N ratio of the simulated systems shows that the non-uniform amplitude and phase (MABF) system is the best one among the other systems because it enhanced the signal to noise ratio by about 5.12 dB above the uniform (MABF) system. While the uniform (MABF) system using four microphones enhanced the S/N ratio by about 3 dB from the uniform (MABF) system using two microphones.
5. The non-uniform amplitude and phase (MABF) system is more difficult and expensive because it needs an adaptive algorithm to obtain the amplitude and phase value through which we get approach nulls in the desired directions.
6. The use of the TMS320C6713 KDS KIT is so suitable for real-time acoustic digital signal processing and so flexible for implementing these systems.
7. The results of the hardware implemented systems indicate the performance of the (MABF) system using four microphones best than the (MABF) system using two microphones. The best system among them is the non-uniform (MABF) system using four microphones this can clearly appear when calculating the S/N ratio for the implemented systems
8. It can be seen from the comparison between the results of the simulation and hardware implementation that there is a small difference of the calculation of the S/N ratio because, for hardware implementation, there is a low level of noise due to the wires and oscilloscope will be added to the system noise.

5.2 SUGGESTIONS FOR FUTURE WORKS:-

1. Test the system in the environment sound room with real sound sources (unrecorded sound sources) and use real microphones with them designed typology.
2. Implement the hardware systems using other equipment like FPGA or other
3. Use another adaptive algorithm instead of (LMS) for calculating the phase and amplitude coefficient in non-uniform beamforming systems
4. Designing new (MABF) systems using more than four microphones and more than four sources.

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ألخلاصة :-

تحسين الكلام هي واحدة من أهم القضايا في مجال معالجة الإشارات والاتصالات. والمعروفة عادة باسم اخماد الضوضاء ألمضافة , ولأن إشارات الكلام غير ثابتة ومتغيرة باستمرار لذلك حتى الآن هناك صعوبات في تعزيز الكلام أو فصله عن الضوضاء الخلفية وإشارات التداخل.

في هذه الرسالة ، تم استخدام تقنية تشكيل مصفوفة المايكروفون (MABF) للحد من الضوضاء وإشارة التداخل وتعزيز إشارات الكلام. تم محاكاة ثلاثة أنواع من نظام (MABF) (نظام متماثل الاتساع والطور (MABF) باستخدام اثنين وأربعة ميكروفونان ونظام غير متماثل الاتساع والطور (MABF) باستخدام أربعة ميكروفونات بواسطة برنامج MATLAB_SIMULINK ومن ثم تنفيذها باستخدام TMS320C6713KDS KIT مع برامج المساعدة. وقد تم استخدام خوارزمية (LMS) لحساب الاتساع والطور لنظام غير المتماثل (MABF)، وقد تم اختبار كل نظام باستخدام إشارات النغمات ومن ثم اختبارها باستخدام إشارات الكلام.

كانت هناك حاجة إلى لوح واحد من معالج الاشارة الرقمية وثلاثة أجهزة كمبيوتر لتنفيذ نظام موحد (MABF) باستخدام اثنين من ألميكروفونات في حين أن تنفيذ النظامين الآخرين يتطلب استخدام اثنين من لوح معالج الاشارة الرقمية وأربعة أجهزة كمبيوتر. تم عرض ومناقشة نتائج جميع الأنظمة المحاكاة والمنفذة عمليا وتم حساب نسبة الإشارة إلى الضوضاء لكل نظام ومقارنتها مع الأنظمة الأخرى.

للأنظمة الثلاثة التي تم استخدامها فان نسبة الاشارة الى الضوضاء تتحسن وبشكل واضح ,حيث ان نسبة الاشارة الى الضوضاء للنظام الاول تساوي 6.18 اما قيمتها للنظام الثاني هي 9.18 بينما النظام الاخير فان هذه النسبة مقدارها 14.3.

وأخيرا ظهر بوضوح أن أفضل نظام تم تنفيذه هو النظام غير المتماثل في الاتساع والطور (MABF) باستخدام أربعة ميكروفونات ويتبعه نظام (MABF) المتماثل باستخدام أربعة ميكروفونات لها نسبة الاشارة المطلوبة \اشارة الضوضاء أقل من غير المتماثل بمقدار 5dB والأخير هو النظام المتماثل (MABF) باستخدام اثنين من الميكروفونات التي لديها نسبة الاشارة المطلوبة \اشارة الضوضاء 3dB أقل من النظام الثاني.

تصميم وتنفيذ نظام تشكيل شعاع الميكروفون

رسالة تقدمت بها

شذى محمد علي

(بكالوريوس هندسة إتصالات)

إلى

مجلس كلية هندسة الالكترونيات في جامعة الموصل

وهي جزء من متطلبات نيل شهادة ماجستير علوم في هندسة الاتصالات

باشراف

الدكتور محمود احمد محمود

1438 هـ   2017م

جامعة الموصل
كلية هندسة الالكترونيات
قسم الاتصالات

تصميم وتنفيذ نظام تشكيل شعاع الميكروفون

شذى محمد علي
(بكالوريوس هندسة إتصالات)

رسالة ماجستير علوم في هندسة الاتصالات

بإشراف
الدكتور محمود احمد محمود

1438 هـ **2017م**