Acoustic Beamforming

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Acoustic Beamforming

85-525 Automotive Applications for Noise, Vibrations and Harshness

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The work throws light on the Acoustic Beamforming fundamentals and advances in the field of beamforming by making sound source localisation more intelligent. The paper describes the arrays design and their utilities for carrying out beamforming measures. It provides a guide to carrying out preliminary computational analysis using the Matlab's "phased array" module under Digital Signal Processing Block. It also draws attention towards the current automotive applications and noise control measures using this technique.

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II. Nomenclature

ANCS	Active Noise Cancellation System
NSI	Noise Source Identification
MSL	Maximum Sidelobe Level
MUSIC	MUltiple SIgnal Classification
TDOA	Time Delay of Arrival
GCC-phat	Generalized Cross Correlation – Phase Transform
DFT	Discrete Fourier Transform
DIT	Decimation in Time
DIF	Decimation in Frequency
FFT	Fast Fourier Transform
SSL	Sound Source Localization
SRP-phat	Steered Response Power – Phase transformation
LCMV	Linear Constrained Minimum Variance
MVDR	Minimum Variance Distortionless Response
DAMAS	Deconvolution Approach for Mapping of Acoustic Sources
PVDF	PolyVinyleDineFluoride
DC	Direct Current
NASA	National Aeronautics and Space Administration

1. Introduction

Acoustic Beamforming is a method of spatial filtering or localisation of the specific sound source from a miscellany of other insignificant sources based on the direction of arrival. The source localisation is governed by the time difference in which the sound reaches the microphone array. Beamforming is very evident in living beings especially animals like dogs where they can localise the source using the brain while hearing to the same.

Acoustic beamforming has wide application spectrum from speech recognition to noise control and detection measures. Acoustic beamforming algorithms are used in conference halls audio video systems where the localisation of the speaker for the camera becomes difficult if done manually [1]. It is used in cell phones/telephones especially when used with the loudspeaker the caller does not hear back his / her voice when the voice is broadcasted via the speaker. It also helps to increase the gain by making the signal very directive so that the signal does not contain any surrounding unwanted noises.

In seismology, to understand behaviours due to reflections of waves on the surface of earth beamforming plays a major role. Beam-forming finds its application in hearing aid design for people facing difficulty with hearing the algorithm and the array system together makes localisation very easy to infer. Robots process the voice commands similarly from a coarser spectrum by localising the source of the command and use it to execute the desired tasks.

The algorithm is used as a part of Active Noise Cancellation System in the car where the car takes the voice instructions irrespective of the presence of background noises and music from the infotainment systems by amplifying or gain to the signal. It also can be used as in-car communication system to reduce the effort of communicating to from and back of the car by the directing the gains to a loudspeaker present closer to the receiver of the information [2]. Beam-forming is used in the fault detector system where the engineer can find the issues in a component present in an automobile or aircraft without actually opening them completely [3]. It is used to monitor the component and the engine health before getting involved in the overhaul process thereby reducing a huge amount of Maintenance Repair and Overhaul timings.

As an inspection tool, it can use to measure the pass-by noise of the vehicles on road [4][5] noise abatement measures in mining areas for the local residents [6]. Thereby helping the designers to develop optimum sound resilient panels and damping accessories to reduce them on the passengers. It also helps to measure the flyby noise characteristics around an airfield thereby regulating the acoustic standards for living beings and infrastructures [7].Using NSI instrumentation the acoustic results are identified to produce visual noise maps which look very similar to the thermal maps but the colors here depict the sound pressure level drops and peaks in the map result [6].

2. Microphone array and applications

The microphones are placed on two-dimensional planes for scanning over a plane or series of planes. There are a lot of ways to place microphones for SSL and beamforming for e.g. sometimes the microphones are placed close to each other to avoid spatial aliasing resulting a higher cost to setup. Irregular spacing of microphones avoids the spatial aliasing with a generic cost. A non-redundant array design has unique inter-microphone spacing.

After exploring all possible methods to deduce a unique pattern researchers found that spiral pattern will be the most effective to design a beamforming microphone array. The arrays are validated by simulations before putting to practice. The beam pattern generated will be able to locate the source at the defined frequency. The bandwidth of the beam varies linearly with frequency. The performance of the filter is measured by the Maximum Sidelobe Level (MSL) where it rejects the unwanted accompanying noises. Deconvolution techniques have recently claimed to remove the effect of array properties and geometry from the beamformed outputs. But the claims are not substantial as still array properties keep influencing the quality of beamformed output [8][9].

2.1 Archimedean Spiral

The Archimedean Spiral obeys the spiral equation given by Archimedes i.e.

$$r(\theta) = a + b\theta$$

a, on changing will rotate the spiral about the center axis.
b, on changing will control the width between the successive turnings. Ref: Fig 8.1 pp 18

To design an archimedean spiral three parameters are necessary minimum and maximum radii ($r_0 and\ r_{max}$) or the microphone position and the angular displacement ($\emptyset$) in radians the spiral should turn to reach. The polar coordinates for the microphones are as follows:.

$$\theta_n = \frac{(n-1)\emptyset}{N-1} \qquad n = 1, 2, 3, \ldots\ldots N$$

$$r_n = \frac{r_{max}-r_0}{\emptyset}\theta_n \qquad n = 1, 2, 3, \ldots\ldots N$$

2.2 Dougherty Log-Spiral

The spiral is designed by placing microphones at equal arc lengths. The Dougherty log-spiral is controlled by maximum and minimum spiral radii ($r_0 and\ r_{max}$), the spiral angle (v) which is the constant angle at which radii from the origin of the spiral are cut by the spiral curve and number of microphones *N*. Ref: Fig 8.2 pp 18

$$l_n = \frac{(n-1)l_{max}}{N-1} \qquad n = 1, 2, 3, \ldots\ldots N$$

$$l_{max} = \frac{r_0\sqrt{1+cot^2(v)}}{\cot(v)}\left(\frac{r_{max}}{r_0} - 1\right)$$

The polar coordinates for the microphones are as follows:

$$\theta_n = \ln\left(1 + \frac{\cot(v) l_n}{r_0\sqrt{1+cot^2(v)}}\right) \quad \text{n = 1, 2, 3,……….N}$$

$$r_n = r_0 e^{\cot(v)\theta_n} \qquad \text{n = 1, 2, 3,……….N}$$

2.3 Arcoundolis Spiral Array

To improve the MSL level at high frequencies the number of microphones is exponentially higher as we move near towards the center of the spiral. The spiral's Cartesian coordinates are controlled by *a* which controls the size of the array; *b*, is how rapidly the spiral expands from the center, $\emptyset$ is the total angle through which the spiral sweeps. ε_x and ε_y are the Cartesian coordinate conversion factors. Ref: Fig 8.3 pp 18

$$a = r_0\left(\frac{N}{1+\varepsilon_x N}\right)$$

$$b = \frac{1}{\emptyset}\ln\left(\frac{r_{max}}{a\sqrt{(1+\varepsilon_x)^2 cos^2\emptyset + (1+\varepsilon_y)^2 sin^2\emptyset}}\right)$$

The Cartesian coordinates for the microphones are as follows:

$$x_n = \left(\frac{n+\varepsilon_x N}{N}\right).\mathrm{a}.\cos(\theta_n).e^{b\theta_n} \qquad \text{n = 1, 2, 3,……….N}$$

$$y_n = \left(\frac{n+\varepsilon_y N}{N}\right).\mathrm{a}.\sin(\theta_n).e^{b\theta_n} \qquad \text{n = 1, 2, 3,……….N}$$

2.4 Multi-spiral array

In this case, there are multiple spiral arms equally rotated about the origin, the spirals can vary in their characteristics but are basically Dougherty Log spiral. The positions of microphones are controlled by maximum and minimum spiral radii ($r_0 and\ r_{max}$), the spiral angle (v) which is the constant angle at which radii from the origin of the spiral are cut by the spiral curve, Number of spiral arms N_a and number of microphones per spiral N_m. Ref: Fig 8.4 pp 18

The polar positions of microphones are determined by using the equations in Dougherty log spiral.

2.5 Underbrink array

The Underbrink array is a modified multi-spiral case where each microphone is placed in the center of the equal area segments. The positions of microphones are controlled by maximum and minimum spiral radii ($r_0 and\ r_{max}$), the spiral angle (v) which is the constant angle at which radii from the origin of the spiral are cut by the spiral curve, Number of spiral arms N_a and number of microphones per spiral N_m. The area under the array is divided into ring like equal area segments with microphones placed at the center of the segments. The inner circle of microphones is added at r_0 to increase the performance of the array under high frequencies. Ref: Fig 8.5 pp 18

$$r_{m,1} = r_0 \qquad m=1,2,3\ldots\ldots\ldots, N_a$$

$$r_{m,n} = r_{max}\sqrt{\frac{2n-3}{2N_r-3}} \qquad n=1,2,3\ldots\ldots\ldots N_m$$

$$\theta_{m,n} = \frac{\ln\left(\frac{r_{m,n}}{r_0}\right)}{\cot(v)} + \frac{m-1}{N_a}2\pi$$

2.6 Brüel & Kjær style array

Brüel & Kjær an acoustic research firm designed an array which is easy to installation and setup for field testing. The setup consists of concentric hoops and spokes connecting the inner hoops to the outer. The spokes are placed at an angle∅. To calculate the microphone locations the number of spokes, N_a, microphones per spoke, N_m, spoke angle, *f*, and a distribution of microphone locations along the spoke: Ref: Fig 8.6 pp 18

$$d_n \in [0\ 1] \qquad n=1,2,3\ldots\ldots\ldots, N_m$$

Microphone radii is given as

$$r_{m,n} = \sqrt{{r_0}^2 + (ld_n)^2 - 2r_0 ld_n\cos(\emptyset)} \qquad m=1,2,3\ldots\ldots\ldots, N_a$$

$$l = r_0\cos(\emptyset) + \sqrt{{r_{max}}^2 - r_0 sin^2(\emptyset)}$$

$$\theta_{m,n} = \sin^{-1}\left(\frac{ld_n}{r_{m,n}}\sin(\emptyset)\right) + \frac{m-1}{N_a}2\pi$$

After analysing the far-field (planar propagation) and near field (spherical propagation) of the arrays, the Underbrink array performs well in resolution and reasonable MSLs when the source is closer to the center and better MSLs when the source is away.

3. Sound source localisation algorithms

3.1 High-resolution spectral estimation (MUSIC)

The MUSIC algorithm (MUltiple SIgnal Classification) is a well-known high-resolution method to sound source localization. For narrowband, several extensions can be imagined when dealing with wideband sources like the human voice and other wide band applications.

3.2 Time difference of arrival information

There are 2 steps for determining the TDOA information

- TDOA estimation of sound signals between two spatially separated microphones.
- Given array geometry and calculated TDOA estimate the 3D location of the source.

Determination of cross-correlation function has to be determined by the TDOA estimation if the signal is stationary the cross-correlation depends on time difference of two signals. A plot between the time lag and time will result in a hyperbolic curve. All the points on the curve act as a source for the sound. And the maxima of the curve give the exact source of the sound.

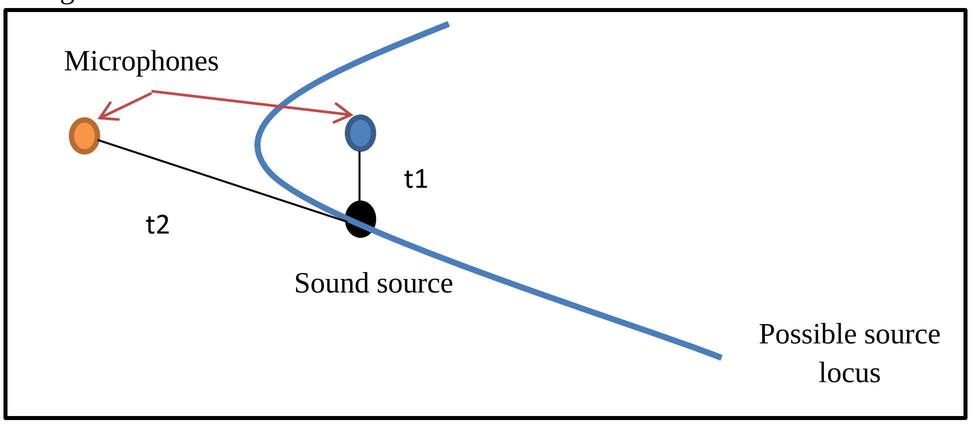


Fig 3.1 Localization using TDOA

Generalized Cross Correlation – Phase Transform: (GCC-PHAT)

Considering a source at point $p(r, \theta, \varphi)$ which is unknown

m_i, is the vector location of microphone 1.

m_j, is the vector location of microphone 2.

The time difference as a function p is known to us as:

$$\tau_{12}\,(p) = \frac{||(p-m_i)||-||(p-m_j)||}{c} \quad \ldots\ldots\ldots.1$$

In which the cross-correlation function is given by:

$$R_{12}(\tau) = \frac{1}{2\pi}\int_0^{2\pi} \psi_{12}(\omega) X_1(\omega) X_2(\omega) e^{j\omega\tau} d\omega \quad \ldots\ldots\ldots.2$$

The weighting function $\psi_{12}(\omega)$ is given by:

$$\psi_{12}(\omega) = \frac{1}{|X_1(\omega) - X_2(\omega)|} \qquad \text{……………3}$$

The cross-correlation function has the maxima for the unique value of τ.

$$\tau_{12} = MAX(R_{12}(\tau)) \qquad \text{…...4}$$

After τ got determined in equation 4, substituting τ in equation 1 to find the exact value of coordinate *p*.

GCC results in multiple peaks in a reverberated environment and makes determination difficult for the source location.

3.3 Steered response power

A beamformer scans, over the predefined region where the sound source is located, at all possible positions. The point giving the maximum beam forming output is the location of the source.

If number of microphones are *m* then Q be its *m*(m-1)*0.5* pairs exist given by C_2^m

Then the steered response power function is given by

$$SRP(p) = 4\pi \sum_{q=1}^{q=Q} R_q(\tau_q(p))$$

The source position is determined by the maxima of the SRP function.

$$\dot{p} = MAX(SRP(p))$$

There is a strong chance of GCC being reverberated resulting a lot of peaks in the signal determining the maxima among them becomes difficult.

The GCC is faster and gives quick real-time results but SRP is a robust grid-based methodology used in critical applications where precision is important.

4. SSL Architecture (Non-Stationary Signals)

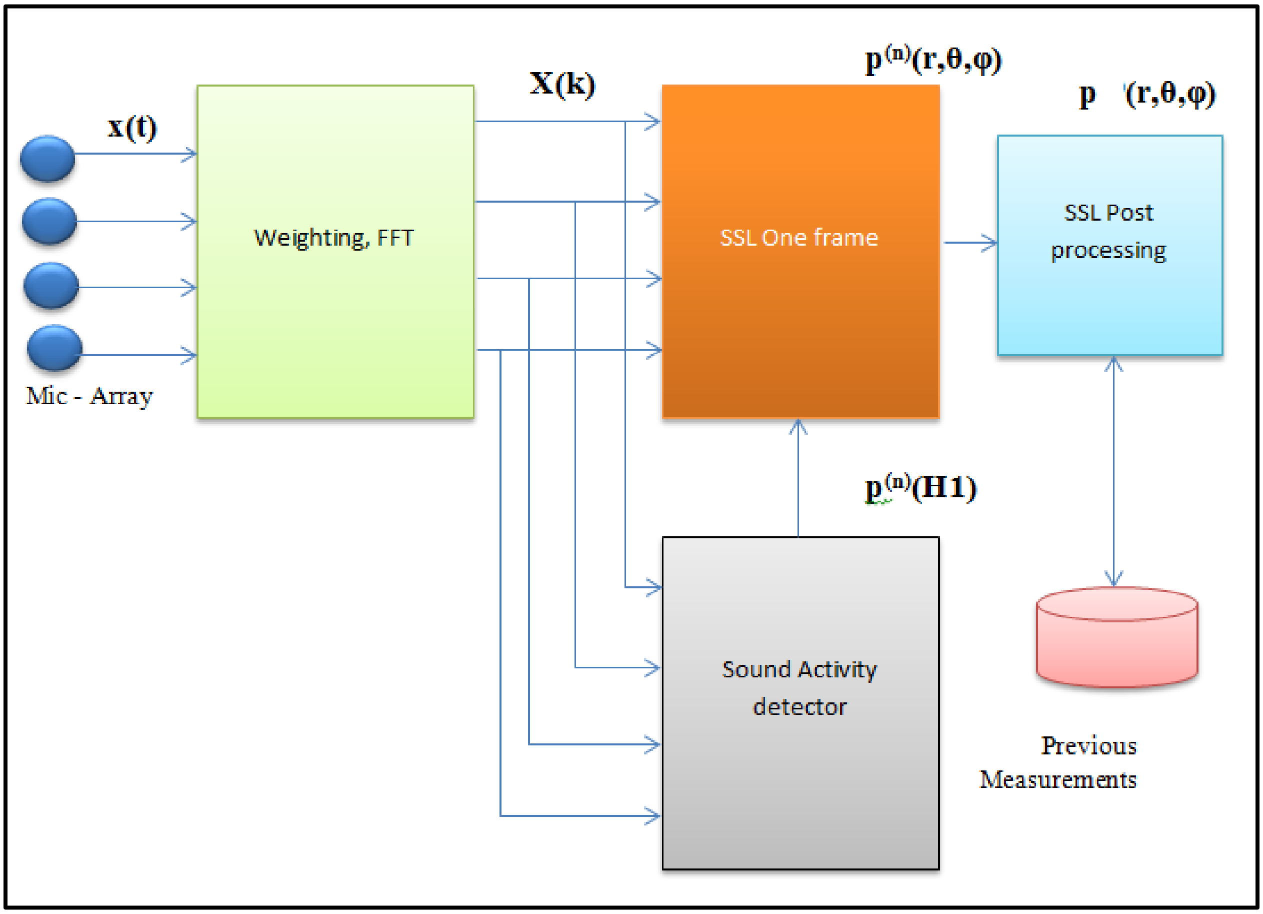


Fig 4.1 SSL Architecture block diagram frequency domain approach

The signals **x(t)** (time domain) picked up from the microphone are analyzed by using pairwise DFTs or DIF based FFT to get **X(k)** (Frequency domain signal) the sound activity detector triggers the SSL one frame localizer to undergo a steered response function maximization on the weighted FFT outputs to determine the source due to the DFT pairs. The output pairs are processed in cross-correlation with the previous measurements.

5. Beamforming

If the location of the source is determined by the GCC-phat, the SRP-phat or the spectral analysis a spatial filtering so called as beamforming is used to prioritize the signals exactly coming from that source to be given as high gain and other regions will be nulled so that we get only the desired signal.

Types of beamforming:

5.1 Delay and sum beamforming

The in this method the time delay for the signal coming from the desired source is taken to account for the steer delay block and all the signals are aligned by introducing a delay in the leading wavefront. After the alignment, it leads to constructive interference of signals thereby amplifying the magnitude of the output waveform from the summation block.

But in the case of the noise sources the correction by introducing the delay is not carried results in a misaligned wave output which on summation does not substantially amplify like the desired source thereby gets filtered from the output easily.

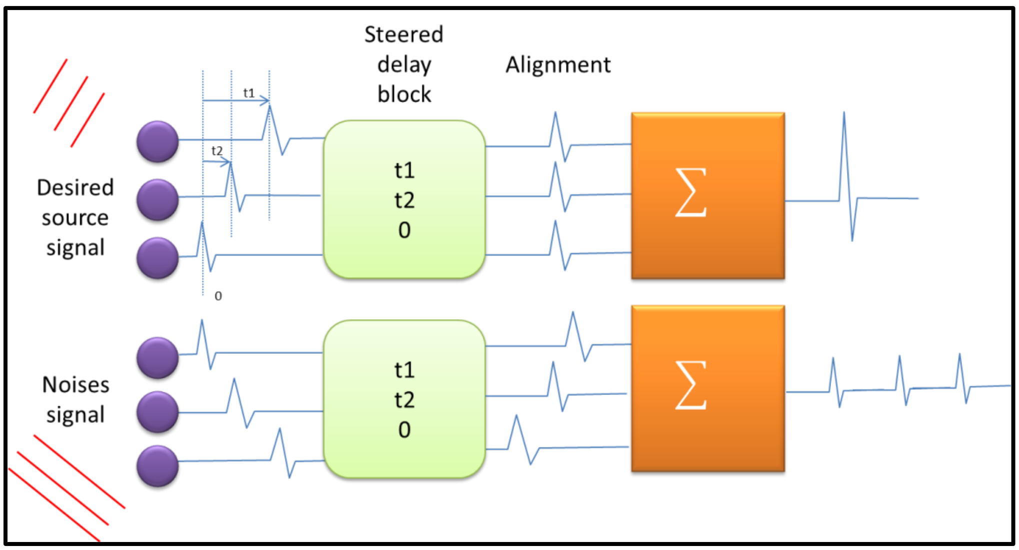


Fig 5.1 Delay and sum beamforming approaches a microphone array time domain

1. Sound signals are multiplied by a weighting factor corresponding to each microphone. This calibrates the microphone to known noise in the test field.

2. An initial sound wave incident angle is chosen and the time delay for each microphone based on that angle is calculated using the following equation:

$$t = \frac{(n-1)d\cos\theta}{c}$$

3. Time delay for each microphone is calculated and the delay is added to the recorded sound which leads to the realignment of the waveforms.

4. Finally summed up to produce a power amplified wavefront.

5. Measurements are taken with all possible angles of incidence to microphone array resulting in a generation of intensity map around the microphone.

Limitations

The drawback using this method is that the degree of resolution that can be able filtered is determined by the sampling rate of your data because delay differences less than your sampling rate cannot be resolved. For example, if the sampling period is 10 milliseconds, then a range of say 20 degrees where the delays would be 5 milliseconds or anything less than 10, resulting in every signal coming from same source. Conditioning signals from anywhere in this range would result in the same signal. [10]

5.1 Linear Constrained Minimum Variance Beamformer

To mitigate the problems encountered using a delay and sum beamformer the LCMV Beamformer minimizes the outputs in form of constraints on the weighting factor. LCMV is an upgraded beamformer of MVDR as it can predict the distance of arrival of the signal thereby improving the SNR.

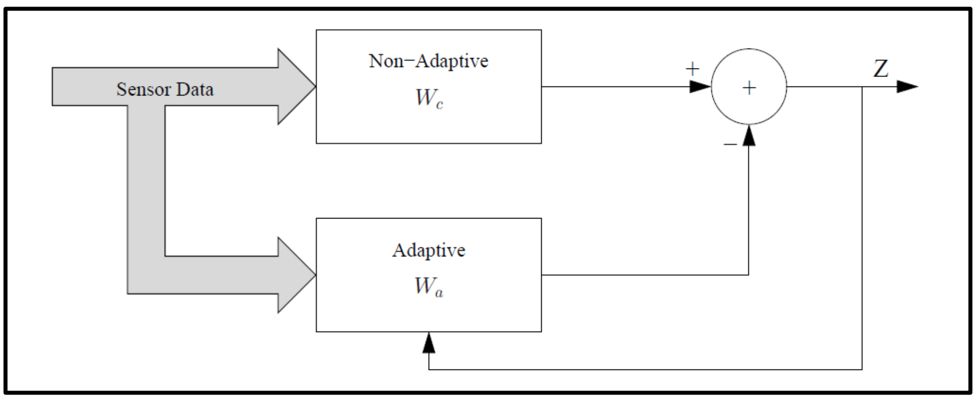


Fig 5.2 The LCMV beamformer dual behaviour an adaptive part and a non-adaptive part. (Adapted from Allred, D. J. (2006). pp. 44)

In LCMV the array outputs are compared with the base signals and beams are being produced in the same direction of multipath signals which matches with the base signals. The mathematics for the weighting as well as signal steering is found in [10]

5.2 Minimum Variance Distortionless Response Beamformer

MVDR Beamformer is adaptive beamformer and falls under the super directive beamforming category. The beamformer is capable of predicting the weighting vectors for beam steering. MVDR can resolve signals separated a fraction of microphone beam width.

MVDR calculates the weight vector to identify the direction for the approaching signal. The beamformer maximizes the sensitivity a particular direction only this is achieved by minimizing the power output of the beamformer on a linear constraint in response to an array of the course of required signal. The use of additional constraints gave rise to the birth of more adaptive beamformers with more constraints and dynamic update of weights which was developed.[11][12] The output response of an MVDR under [17] is given in the Fig 8.7 and Fig 8.8 (pp 19) gives the comparative power response of both LCMV and MVDR for the same signal.

5.3 Generalized side lobe canceller beamformer

In the case of this method of beamforming, the arrays are categorized into auxiliary and main types. The main array looks to source localizations whereas the auxiliary one takes care of the interferences. The interferences are subtracted from the main array output to provide a refined result. While using a side-lobe beamformer it was observed that the auxiliary microphones also pick some of the desired signals which lead to a kid of distortion in the final output. To mitigate the issues Jim and Griffith [9]came out with a concept of blocking matrix which stops/blocks the propagating of the primary signals to the auxiliary arrays. The block diagram in Fig differentiates a simple sidelobe canceller and a generalized side lobe canceller.

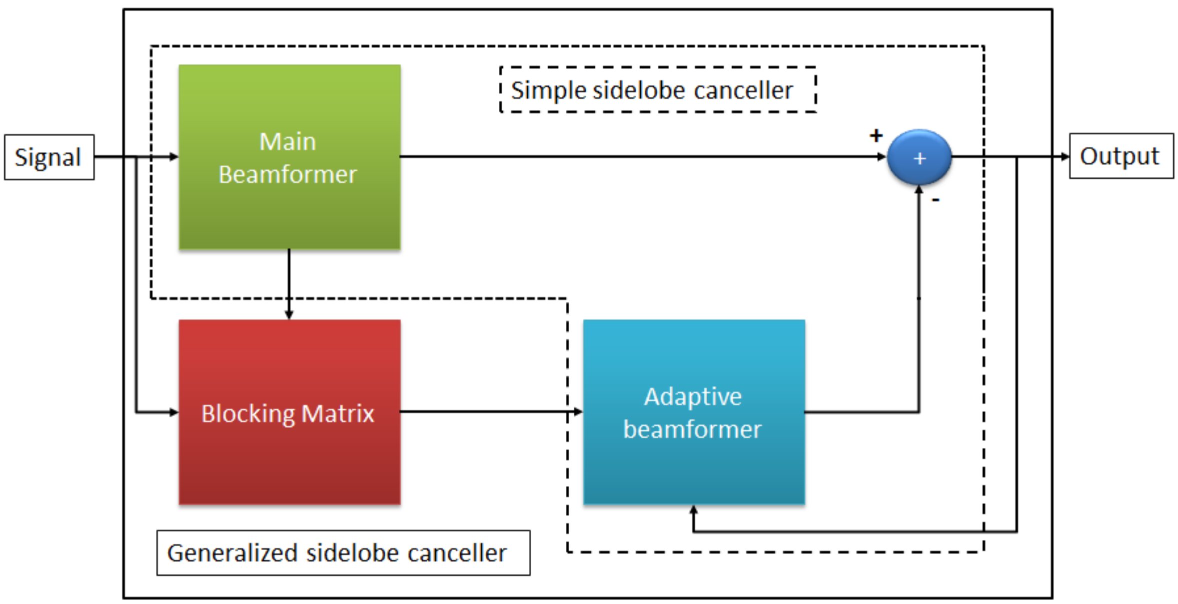


Fig 5.3 Generalized sidelobe canceller block diagram

5.4 Conclusion

To refine the beamformer to the as per the task complications led to a development of large no of algorithms which improves the modularity and adds intelligence to the task of sound source localization. A few more popular beamformers include the Frost beamformer [14], Griffith and Jim beamformer, DAMAS approach to refine beamforming [9]. To solve the problem of calibration NASA together with the Michigan State University that self-calibrates the microphones using closed-loop feedback control [15]. Manual calibration consumes a lot of time and may need to be repeated regularly to take into account effects such as heat and humidity. A thin polyvinyledinefluoride (PVDF) film membrane, which allows for actuation in a feedback scheme. The feedback actively modifies such sensor parameters as membrane stiffness, resonant frequency, damping, and DC attenuation. A fiber-optic sensor was used to detect displacement of the membrane's center caused by acoustic pressure Ref Fig 8.11 pp 20.

6. Computational Analysis in Matlab

The Matlab has a special toolbox which deals with the array of microphones called Phased Array System Toolbox. The toolbox contains pulsed and continuous waveforms and signal processing algorithms for direction of arrival (DOA) estimation, beamforming, matched filtering and target detection. To simulate the parameters in a real-time scenario it has mathematical models for transmitters and receivers, propagation, targets, jammers, and clutter.

Using Matlab you can infer the beam patterns of the discussed arrays as 2.1 we can see that the coordinates of the microphones in the array are determined by the polar relations. With the help of those relations, the mathematical model of the arrays can be determined easily. These models can be directly used to establish which performs the best in different scenarios.

For instance:

To create a microphone element

Mic1 = phased.CustomMicrophoneElement;

Assigning frequencies to the microphone should be able to measure

Mic1.PolarPatternFrequencies = [500 1000];

Specifying the measuring angles in degrees of the polar patterns as a row vector of length N

Mic1.PolarPatternAngles = [-180 180];

Adding microphone pattern details if M is the number of frequencies specified in the PolarPatternFrequencies property and N is the number of measuring angles specified in the PolarPatternAngles property. Each row of the matrix represents the magnitude of the polar pattern (in decibels) measured at the corresponding frequency specified in the PolarPatternFrequencies property and corresponding angles specified in the PolarPatternAngles property. The pattern is considered for a cardioid microphone.

Mic1.PolarPattern =
mag2db([0.5+0.5*cosd(Mic1.PolarPatternAngles);0.6+0.4*cosd(Mic1.PolarPatternAngles)]);

To determine the output response of the microphone under those frequencies

Resp = step(Mic1,[500 1500 2000],[0 0;40 50]’)

The Resp is an M-by-N matrix that contains the responses of the microphone element at the N angles specified in [0 0; 40 50]^{c} (complex conjugate)and the M frequencies specified in [500 1500 2000 10000 20000].

To plot the response

plotResponse(Mic1, [500 1500],'RespCut', 'Az','Format','Polar');

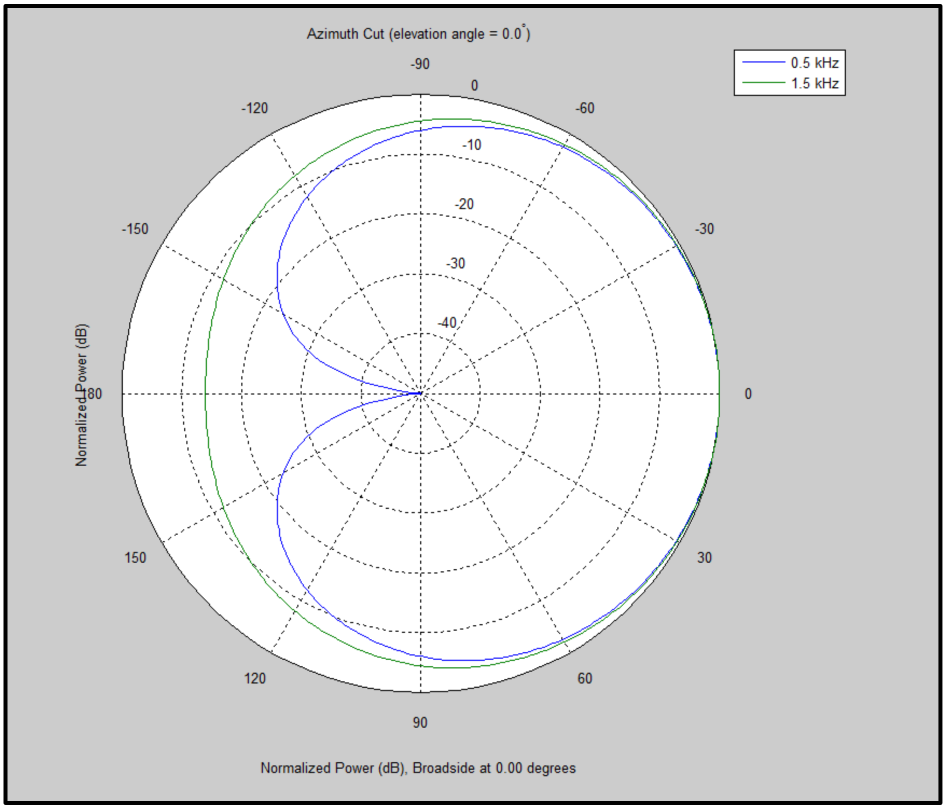


Fig 6.1 Response pattern of cardioid microphone generated for frequencies at different angles

Similarly, multiple microphones (Mic2, Mic3, Mic4, Mic5 …. MicN) can be created and spiral arrays can be modelled using Matlab.

The algorithm for beamforming using a 2 D microphone array can be given as

- Determine the Position of sensors and weighting of 2D array
  - Inputs are number of elements, radius of the array, polar coordinate values
  - Outputs are the Cartesian x,y,z and weights
- Defining the arriving angles of input signals (theta and phi)
- Defining the Array scanning angles (theta and phi)
- Generation of input signal
  - Inputs are x,y,z, the frequency of the signal, speed of sound, arrival angles(theta and phi).
  - The output is Signal.
- Implementation of the beamforming algorithm using the scanned signal data with the x,y,z, frequency of the signal and speed of sound.

7. Automotive applications of Acoustic beamforming

Pass by the noise of automobiles is a huge concern for acoustic developments around a vehicle. They majorly monitored by the federal laws who define ambient noise standards for highways and the vicinity. The concept of beamforming adds more depth by identifying the source or the exact region responsible for producing that noise. Beam forming is effective when the waves are flat and the source must be in a far-field region rather than in a near field or reverberant region.

7.1 International standards governing the pass by noise calculation.

ISO has defined these standards for measuring the noise sources

- ISO 362 – pass by noise using a single microphone.
- ISO 13325 – Tire Noise determination
- ISO 5130 – Exhaust noise
- ISO 10844 – it defines the test track for the carrying out the noise tests on automobiles

Quoted from the ISO 10844:2014 en [16] "The surface design given in this International Standard

- produces consistent levels of tire or road sound emission under a wide range of operating conditions including those appropriate to vehicle sound testing,
- minimises inter-site variation,
- provides minor absorption of the vehicle sound sources, and
- is consistent with road-building practice."

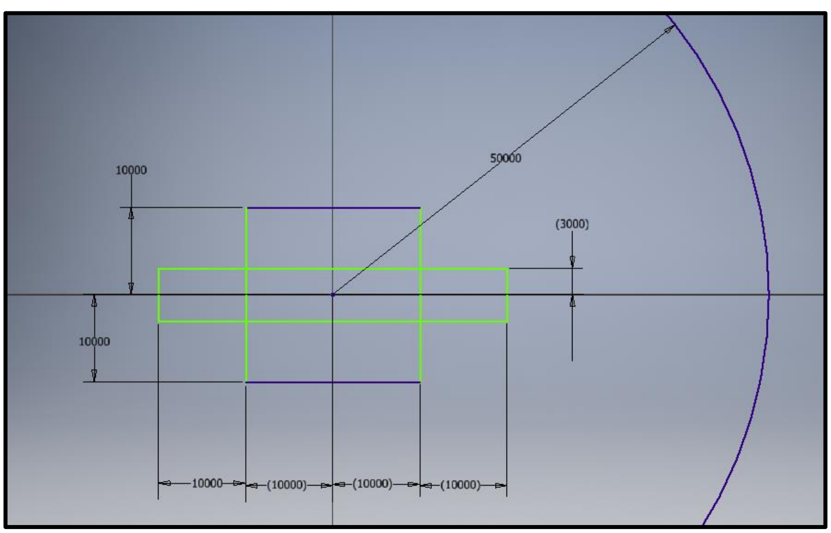


Fig. 7.1 ISO 10844 specifies test tracks for vehicle noise emissions tests. The dimensions are in millimetres (reproduced exactly from the specified ISO 10844 test track specification)

In any beamforming or sound measurement, the resolution and dynamic range of the measurements play a huge role. Both of these parameters are controlled by the onboard hardware such as microphones and signal conditioning/data acquisition hardware.

Resolution in acoustic beamforming is the spatial resolution which is given by the product of the wavelength of the sound and ratio between the distance from the source and diameter of the array. [5]

$$SR = \frac{Distance\ of\ the\ source}{Diameter\ of\ the\ array} \times sound\ wavelength$$

Whereas the **dynamic range** is the ability to differentiate a sound from noise environment around it i.e. a high dynamic range can easily identify the desired sound from a noisy environment. Again it is completely dependent upon the microphone arrangement and the sensitivity of the microphones in general.

Fig 7.2 To visualize exact location of the noise source a camera is integrated with the microphone array by superimposing the intensity spectrum on the image/video. (Adapted from Farrell, D)

8. Appendices

8.1 Array designs(Prime & Doolan, 2013, pp. 1-7)

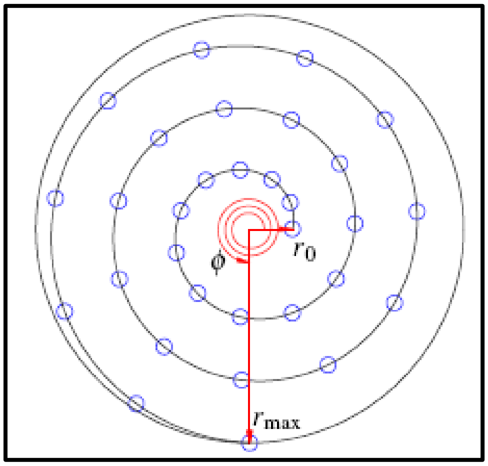


Fig 8.1 Archimedean spiral array design

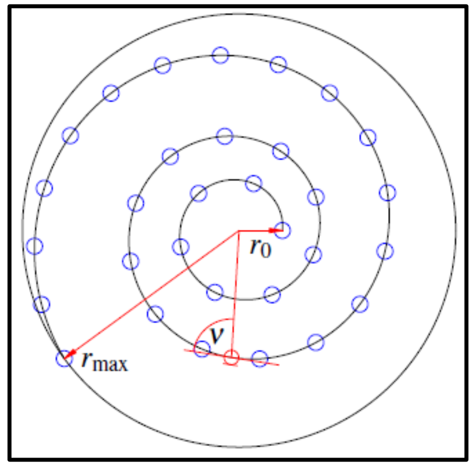


Fig 8.2 Dougherty log-spiral design

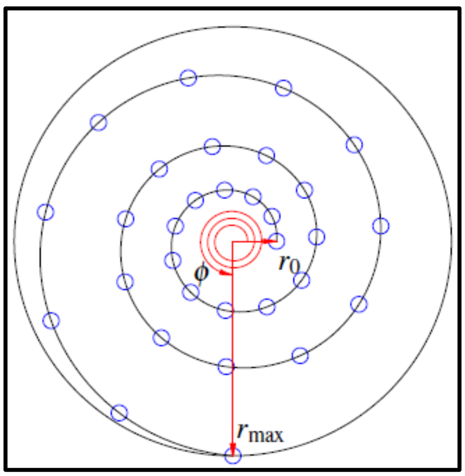


Fig 8.3 Arcondoulis spiral design

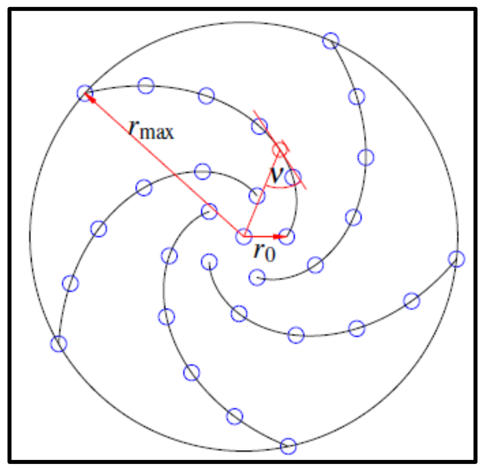


Fig 8.4 Multispiral design

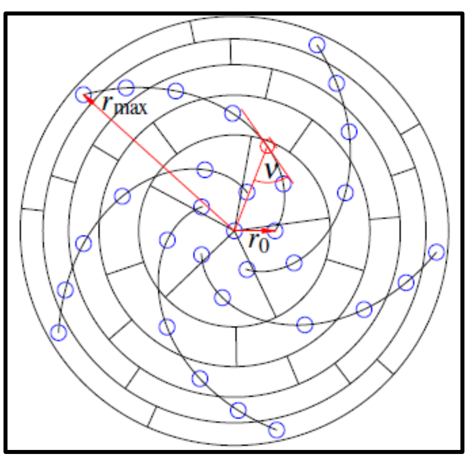


Fig 8.5 Underbrink spiral design

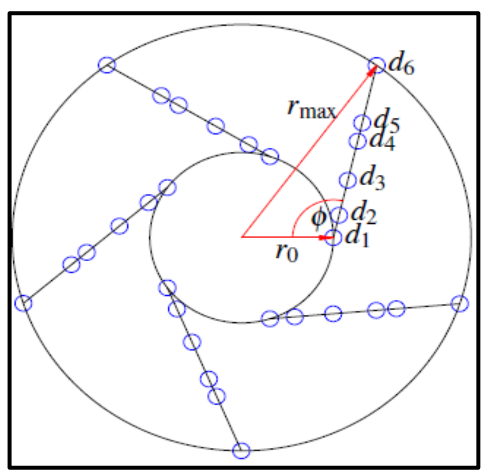


Fig 8.6 B&K array design

8.2 MVDR and LCMV Plots

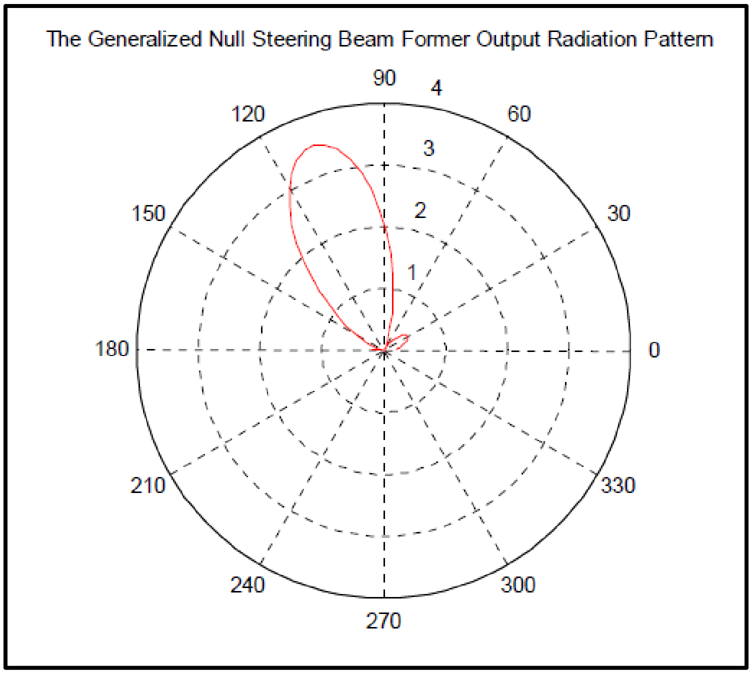


Fig 8.7 Polar Plot of MVDR Beamforming (Balasem, Koh, Tiong, 2012 pp 322)

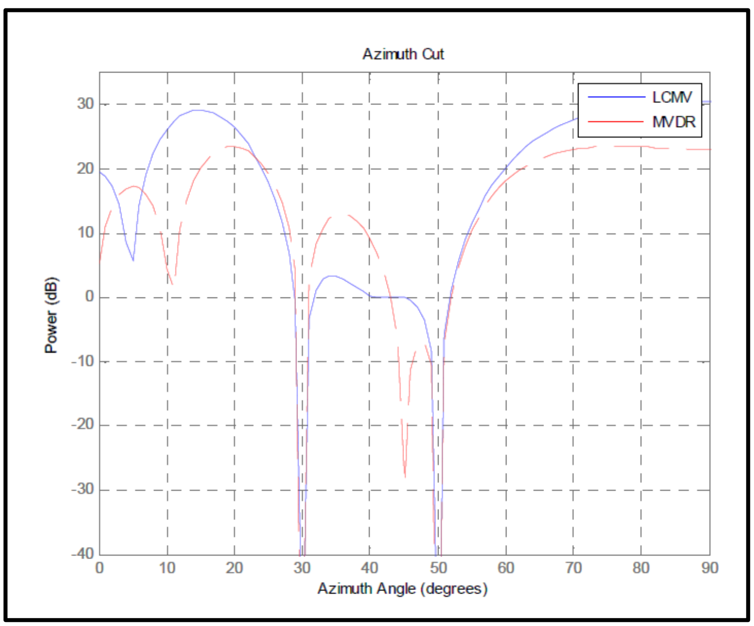


Fig 8.8 LCMV vs MVDR Beamformer (Balasem, Koh, Tiong, 2012 pp 324)

8.3 Experimental Demonstrations

Fig 8.9 – Bruel & Kjaer penta-angular microphone array setup for pass-by measurements. (Adapted from Angione, Frank, 2015)

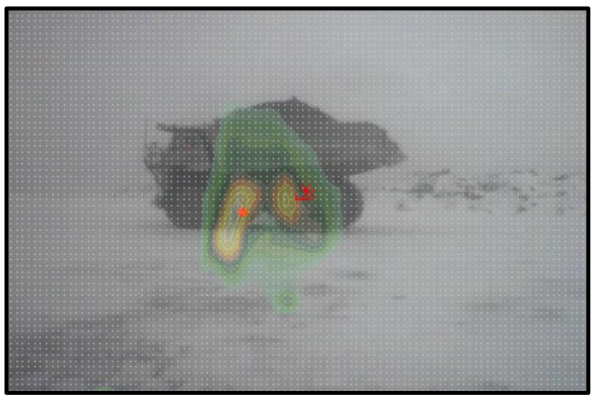
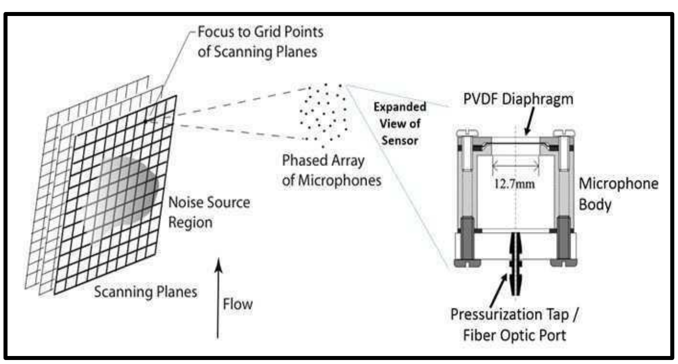


Fig 8.10 – A-weighted sound intensity level noise map from the driver's side of the 785B mining haul truck with aftermarket acoustical louvre. (Adapted from Angione, Frank, 2015)



(a) (b)

Fig 8.11- (a)Microphone Phased Array Operation with Sensor Details (b) Setup at an Anechoic room (Adapted from Radcliffe, E., Naguib, A., & Humphreys, W. M. (Sep 30, 2014))

9. References

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