Blind Beamforming on a Randomly Distributed Sensor Array System

A Thesis Submitted in Partial Fulfilment of the Requirements for the Degree of Bachelor of Technology in Electronics and Instrumentation Engineering

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Declaration

I hereby declare that the project work entitled "**Blind Beamforming on a Randomly Distributed Sensor Array System**" is a record of my work done under Professor Lakshi Prosad Roy, National Institute of Technology, Rourkela. Throughout this documentation wherever contributions of others are involved, every endeavour was made to acknowledge this clearly with due reference to literature. This work is being submitted in the partial fulfilment of the requirements for the degree of Bachelor of Technology in Electronics and Instrumentation Engineering at National Institute of Technology, Rourkela for the academic session 2011-2015.

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Certificate

This is to certify that the thesis entitled "**Blind Beamforming on a Randomly Distributed Sensor Array System** "submitted by Anshuman Jena(111EI0244) in partial fulfilment of the requirements for the award of Bachelor of Technology Degree in Electronics and Communication Engineering at National Institute of Technology, Rourkela is the work carried out by him under my supervision and guidance.

Prof. Lakshi Prosad Roy
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Acknowledgement

This is a research project and the fact that I have been able to complete it successfully owes a lot to a number of persons associated with me during this project.

First of all, I would like to thank Prof. Lakshi Prosad Roy for giving us a golden opportunity to work on such an interesting topic and providing a thoroughly professional and research oriented environment. He also guided me nicely throughout the project period and helped me time to time with his vast experience and innovative ideas.

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Abstarct

We consider a digital signal handling sensor array system, in light of haphazardly dispersed sensor node, for observation and source localization applications. In most array handling system, the sensor array geometry is settled and known and the steering array vector/complex data is utilized as a part of beam- formation. In this system, the array adjustment may be illogical because of obscure situation and introduction of the sensors with obscure frequency/spatial responses.In this project work a blind beamforming method is used by utilizing just the deliberate sensor information, to shape either an example information or a sample correlation matrix. The greatest power accumulation measure is utilized to acquire array weights from the predominant eigenvector connected with the largest eigenvalue of a matrix eigenvalue issue. A productive blind beamforming time delay appraisal of the predominant source is proposed. Source localization in light of a least squares (LS) technique for time delay estimation is additionally given. Results taking into account investigation, simulation, and measured acoustical sensor information demonstrate the viability of this beamforming system for sign upgrade and spacetime filtering.

Contents

List of Figures

1 Introduction

In the recent 20 years, there has been much enthusiasm for the hypothetical and handy parts of beamforming. Beamforming has been utilized as a part of radar, sonar, and wireless radio applications. The sources may be narrowband or broadband in the far-field or close field as per the application, there may be multipath or reverberant echoes, and the recurrence furthermore, spatial reactions of the sensors may be totally or part of the way obscure. The issue of beamforming in a narrowband domain, with absolutely known sensor areas and reactions, is well archived. Distinctive calculations use the structure of the directing grid to acquire data with respect to source heading of landing. Numerous high-determination bearing of landing estimation calculations have been proposed lately, yet none of them are suited to un-calibrated arbitrarily appropriated sensors clusters, inactively getting wideband signals. MUSIC [1] and ESPRIT [2] oblige narrowband signals. An impressive measure of examination has been committed to the troublesome issues of beamforming in the vicinity of rational sources and the impacts of imprecisions in the sensor calibration furthermore, area have likewise been examined.

Beamforming for broadband sources has more often than not been considered as an augmentation of narrowband beamforming in the recurrence area by utilization of sub-band sifting and/or centering framework methods. This technique likewise obliges sensor reaction calibration. At the point when the data in regards to sensor arrangement and reaction is incompletely or totally inadequate with regards to, the beamforming issue is normally alluded to as visually impaired beamforming. Various articles have managed this topic, typically in the zone of advanced correspondences. The normal situation includes narrowband wellsprings of which some known attributes are utilized for the motivation behind identification or signal duplicate. Among the highlights misused are: the cyclostationarity property [3], other worldly self-coherence [4] or the limited letter set property of advanced correspondence signals, the consistent modulus characteristic of frequency modulation/phase modulation (FM/PM) [5] signals, the measurable contrast in the middle of craved and undesired sources, including sorts of signal nonstationarity [6], and higher request measurable parameters. The last class of issues has created a wide mixture of articles in which higher request cumulants have been adequately used to battle the impact of mesokurtic aggravations, for example, Gaussian noise [7] .

There has been much late enthusiasm for utilizing low-power and low-cost integral metal-oxide-semi-conductor (CMOS) created miniaturized micro electromechanical (MEM) sensors, in conjunction with advanced computerized signal processors (DSP's) and radio recurrence (RF) radio correspondence methods, to tackle different testing issues including the coupling of information from the physical world through a system to the end client. In this paper, we consider a show framework in which the sensor hubs may be haphazardly disseminated. The client may have the control of some broad parameters of situation of these hubs, for example, the inexact thickness of the hubs, and a surmised one-dimensional example versus a two-dimensional zone sending. The careful arrangement, control of introduction, what's more, information of recurrence/spatial reactions of the sensors, in any case, are for the most part thought to be unreasonable. These hubs may contain acoustical, vibrational, and other MEM sensing components. These hubs, after sensing an occasion of interest, might self-compose into a synchronized wireless radio system utilizing low-power spread spectrum transceivers to convey among themselves and focal processors. Information from these hubs may be utilized to perform different agreeable signal transforming and beamforming operations for recognition reconfirmation; to lessen the likelihood of false caution; source restriction, and signal-tonoise ratio (SNR) improvement for source signature recognizable proof, and so on.

2 Beamforming

The most broadly utilized array preparing technique is called beamforming. It alludes to any strategy that algorithmically directs the sensor in the array towards target signal. The course is called look heading. Self-assertively set sensor can act as an amplifier array and results in beamforming of acoustic signal. The total of every individual component is yield from a receiver array. In such case the yield from every individual amplifier is postponed rendition of source flag and lessened by a factor i (i speaks to ith mouthpiece) furthermore it contains some uncorrelated noise.so yield from every receiver is

$$x_i(t) = a_i s_i(t - \tau_i) + v_i(t) \tag{1}$$

By taking Fourier transform we get

$$X_i(f) = a_i s_i e^{-j2\pi f \tau_i} + V_i(f) \tag{2}$$

The term A (f, r) is known as the aperture function or the affectability function, and it characterizes the reaction as a function of spatial position along the aperture. The reaction of a getting aperture is naturally directional in nature, on the grounds that the measure of sign seen by the aperture changes with the course of entry. The aperture reaction as a function of frequency and heading of landing is known as the aperture directivity pattern or shaft pattern. To improve aperture directivity we take direct aperture of length L on x hub so spatial area r is disentangled.

2.1 Types of Beamforming

There are distinctive sorts of beamforming procedures. We pick any of them as per our prerequisite and they are examined underneath.

2.1.1 Fixed Beamforming

Fixed beamforming is used when source is not moving and in this case weights of the filters are fixed so desired look direction is always same and examples of fixed beamforming are delay-sum-beamformer and filter sum beamformer which are discussed below.

2.1.2 Delay Sum Beamforming

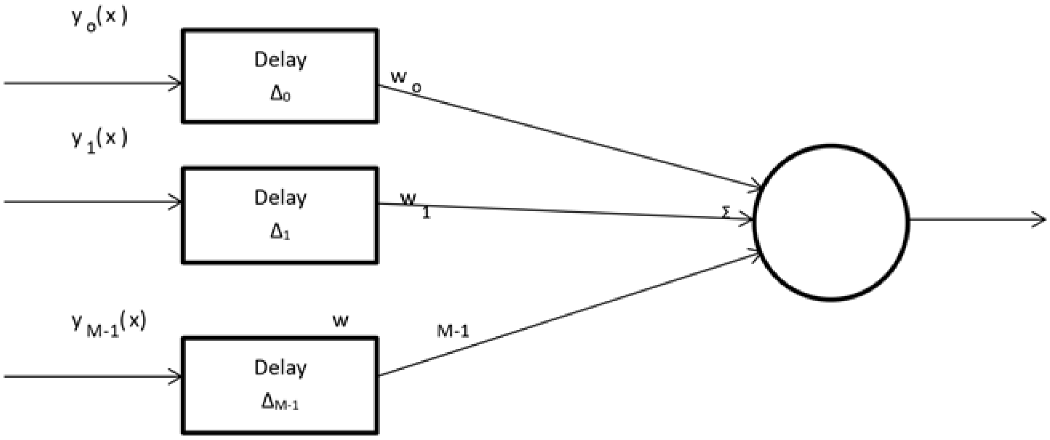

Figure 1: Delay Sum Beamformer

In deferral and-aggregate, signals from the different mouthpieces are first time adjusted to modify for the deferrals created by the way length contrasts between the target source and each of the amplifiers, utilizing an assortment of routines. The adjusted signs are then summed together. Any meddling clamour sources that don't lie along the look bearing stay misaligned and are constricted by the averaging. To adjust, every divert in common time reference, a mouthpiece area is normally chosen as a kind of perspective point. In frequency area, deferral sign is gotten by applying a stage movement to every channel signal range. To standardize yield after summation the sign channel is scaled by uniform addition figure that is number of receivers N and channel weight for deferral aggregate.

Numerous mouthpiece array-based discourse acknowledgment frameworks have effectively utilized postpone and-entirety transforming to enhance acknowledgment execution, and as a result of its straight forwardness, it remains the "strategy for decision" for some array based frameworks. Most other array transforming systems are varieties of this fundamental plan or its characteristic expansion, channel and-total preparing, where every mouthpiece channel has a related channel and the caught signs are initially separated before being consolidated.

2.1.3 Adaptive Beamforming

In adaptive beamforming, the array-processing parameters are dynamically changed according to some optimization criterion. In moving speaker environment where delay between two microphones changes we need to change weights according to surrounding changes or speaker position in those cases we use adaptive beamforming technique. The Frost algorithm is a weighted delay-and-sum technique in which the weights applied to each signal in the array are adaptively adjusted using constrained least mean square (LSM). Another method is the Griffiths-Jim algorithm or generalized side lobe canceller (GSC) which uses minimum variance distortion less response (MVDR). In some cases parameters are calibrated according to particular environment or user. In such cases noise reduction is updated.

2.1.4 DE Reverberation Technique

Resonation is a primary driver of poor speech acknowledgment execution in amplifier array-based speech acknowledgment frameworks. Since none of the conventional beam framing systems effectively adjust for the negative impacts of resonation on the speech signal. So here we estimate room attributes in which we utilize amplifier. As room reaction brings about non-least stage it is hard to gauge definite room reaction.

2.2 Maximum Power Collection Array

In this case the sensor are randomly placed in a spatial region. For beam formation the position and frequency of sensor are unknown. Narrowband or broadband signal are used depending up on far or near field of the sensor array. Due to the distance reverberation and echo may occur. So our initial goal is to the detection, enhancement, and relative-time delay estimation of signal from the noise. If the location of sensors are known we can calculate the time delay to locate the strong signal with respect to the sensor.

The input waveform at the r^{th} sensor ($r = 1,2,3,....,R$) is denoted by

$$x_n(t) = \sum_{d=1}^{D} s_d(t - t_{d,r}) + n_r(t) \tag{3}$$

where, $t_{d,r}$ is the propagation time from the d^{th} source to the r^{th} sensor & $n_r(t)$ is a temporarily and spatially white noise.

The sensor data vector at the three sensors and their combined sampled vectors is given as follows.

$$x_1 = [x(n), x(n-1), ..., x(n-L+1)]^T \tag{4}$$

$$x_2 = [x(n-p), x(n-p-1), ..., x(n-p-L+1)]^T \tag{5}$$

$$x_3 = [x(n-q), x(n-q-1), ..., x(n-q-L+1)]^T \tag{6}$$

$$x = \left[x_1^T, x_2^T, x_3^T\right]^T \tag{7}$$

where, $p = t_{12}$ is the relative time delay of the first sensor to the second sensor & $q = t_{13}$ is the relative time delay of the first sensor to the third sensor.

The auto and cross-correlation matrices of x_1 with x_1, x_1 with x_2, x_1 with x_3, x_2 with x_3, and x with x are respectively given by

$$R_L^{11} = E\left\{x_1 x_1^H\right\} = E\left\{x_2 x_2^H\right\} = E\left\{x_3 x_3^H\right\} \tag{8}$$

$$R_L^{12} = E\left\{x_1 x_2^H\right\} \tag{9}$$

$$R_L^{13} = E\left\{x_1 x_3^H\right\} \tag{10}$$

$$R_L^{23} = E\left\{x_2 x_3^H\right\} \tag{11}$$

$$R_L^{3L} = E\left\{x x^H\right\} = \begin{bmatrix} R_L^{11} & R_L^{12} & R_L^{13} \\ R_L^{12H} & R_L^{11} & R_L^{23} \\ R_L^{13H} & R_L^{23H} & R_L^{11} \end{bmatrix} \tag{12}$$

where, H is the complex conjugate transpose.

The ouput of beamformer is given by

$$y(n) = \sum_{r=1}^{R} \sum_{l=0}^{L-1} w_{rl}^* x_r(n-l) \tag{13}$$

where, w_{rl} is the l^{th} array weight coefficient of the r^{th} sensor.

3 Types of Signal

1. Narrow Band Signal

2. Broad Band Signal

3.1 Narrow Band Signal

Bandwidth of message doesn‘t significantly exceed channel‘s coherence bandwidth. Its frequency response is considered flat. Used in audio spectrum to describe sounds with narrow band width. It is used for far field.

3.2 Broad Band Signal

Simultaneously transports multiple signals and traffic types. Medium can be co-axial cables, optical fiber twisted pair, or wireless broadband. It is used for near field.

4 Speech Recognition

Speech recognition is the process of recognizing who is speaking rather than what the speaker speaking is based on the information/data stored. This is done in two stages namely learning stage and the testing stage. In the training stage the speaker has to utter something to feed the data as speech samples. And in the testing phase the input is matched with the sample to validate the speaker. This process helps to authenticate or verify the identity of a person.

The workflow consists of three steps:

1. Acquiring
2. Analysis
3. Interface Development

4.1 Acquiring Speech

For training, speech is acquired from a microphone and brought into the development environment for analysis. For testing, speech is continuously streamed for processing. During the training stage, it is necessary to record repeated utterances of each digit in the dictionary. For example, we repeat the word one many times with a pause between each utterance. Using the following MATLAB code with a standard PC sound card, we capture ten seconds of speech from a microphone input at 8000 samples per second. We save the data to disk as mywavefile.wav. This approach works well for training data. In the testing stage, however, we need to continuously acquire and buffer speech samples, and at the same time, process he incoming speech frame by frame, or in continuous groups of samples. The MATLAB code shown uses a Windows sound card to capture data at a sampling rate of 8000 Hz. Data is acquired and processed in frames of 80 samples. The process continues until the RUNNING flag is set to zero.

4.2 Analysing the Acquired Speech

Starting with a word-detection algorithm that separates each word from ambient noise. Then deriving an acoustic model that gives a robust representation of each word at the training stage. Finally, we select an appropriate classification algorithm for the testing stage.

5 Developing a Speech-Detection Algorithm

The speech-detection algorithm is developed by processing the pre-recorded speech frame by frame within a simple loop. For example, the MATLAB code continuously reads 160 sample frames from the data in speech. To detect isolated digits, we use a combination of signal energy and zero-crossing counts for each speech frame. Signal energy works well for detecting voiced signals, while zero-crossing counts work well for detecting unvoiced signals. Calculating these metrics is simple using core MATLAB mathematical and logical operators. To avoid identifying ambient noise as speech, we assume that each isolated word will last at least 25 milliseconds.

6 Sound Source Localization

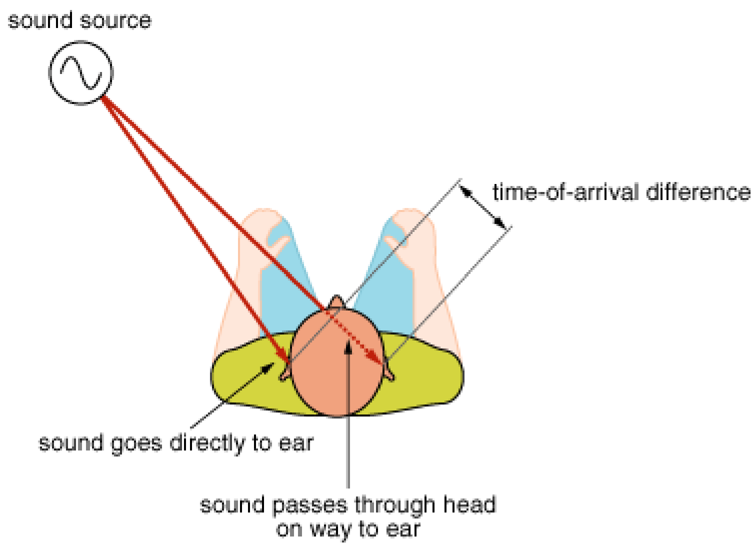

Figure 2: Sound Source Localization

It is a new technique for detection of source from which the signal is coming. Now a days it is basically used in humanoid robot. Four sensor are placed two on two ears and other two is placed on temporal side of the robot. The sensor detect the signal received in the shortest time among all the frequency from the source. It decodes the signals with an accuracy up to 20 degrees and locates the source with relatively higher efficiency. So that the robot can move his head towards the source and the sensor placed on two eye can detect the source.

7 Working of Filters

Basically filter are used to filter the audio signal of desire frequency range. Mainly it is of two type FIR (finite impulse response) and IIR (infinite impulse response). Within this category there are so many filter like high pass filter, low pass filter, band pass filter, band reject filter, all pass filter etc.

The measure of weakening for every frequency relies on upon the filter plan.The filter is some of the time called a high-cut filter, or treble cut filter in sound applications. A low-pass filter is the inverse of a high-pass filter. A band-pass filter is a mix of a low-pass and a high-pass filter

7.1 High Pass

It pass the frequency of signal above the cut-off frequency and stop the other frequency.

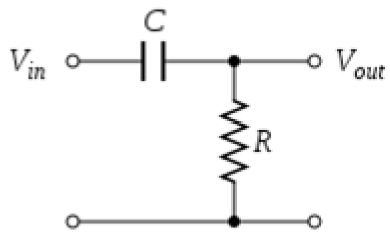

Figure 3: High Pass Filter

A high-pass filter is normally demonstrated as a linear time-invariant system. It is sometimes called a low-cut filter or bass-cut filter. High-pass filters have numerous uses, for example, blocking DC from circuitry sensitive to non-zero average voltages or radio frequency gadgets. They can likewise be utilized as a part of conjunction with a low-pass filter to deliver a band pass filter.

7.2 Low Pass

It pass the frequency of signal below the cut-off frequency and stop the other frequency.

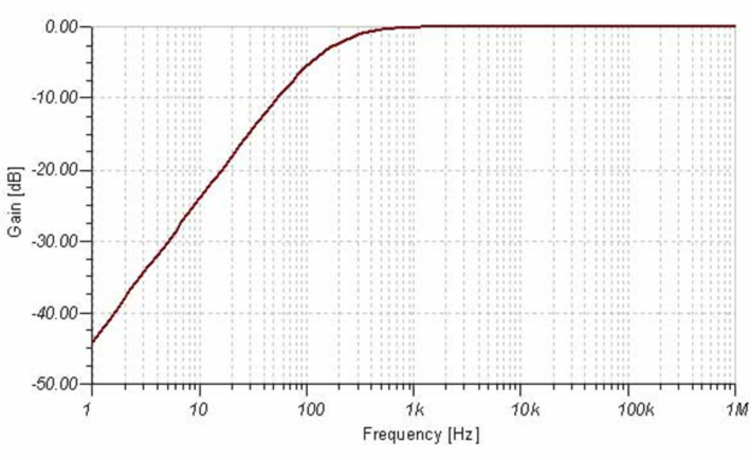

Figure 4: High Pass Filter Frequency Response

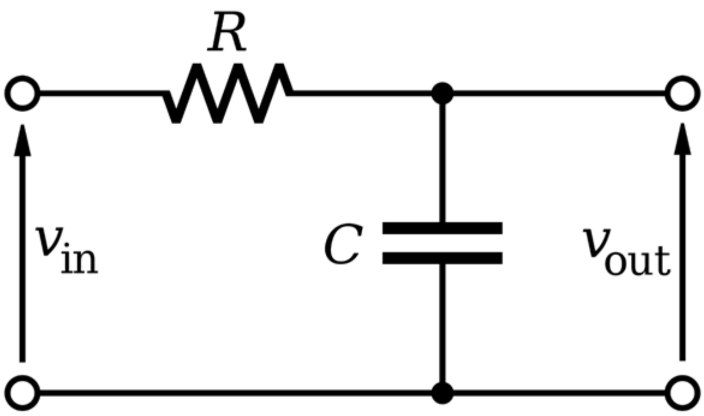

Figure 5: Low Pass Filter

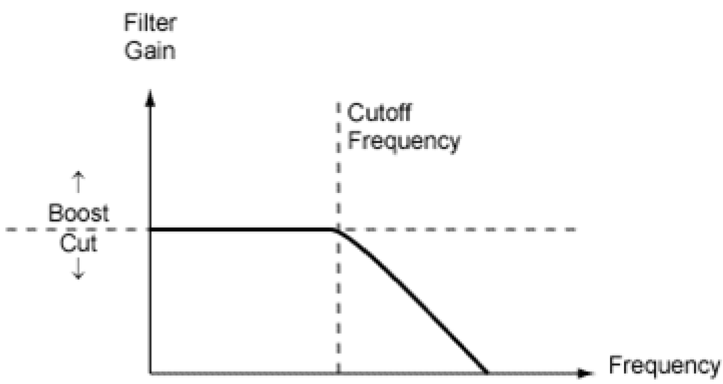

Figure 6: Low Pass Filter Frequency Response

7.3 Band Pass

It pass the frequency of desire band and reject other frequency.

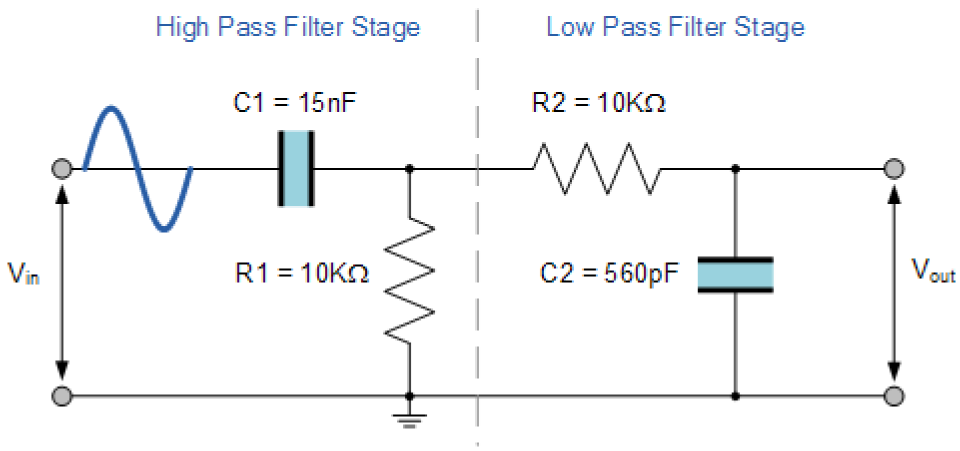

Figure 7: Band Pass Filter

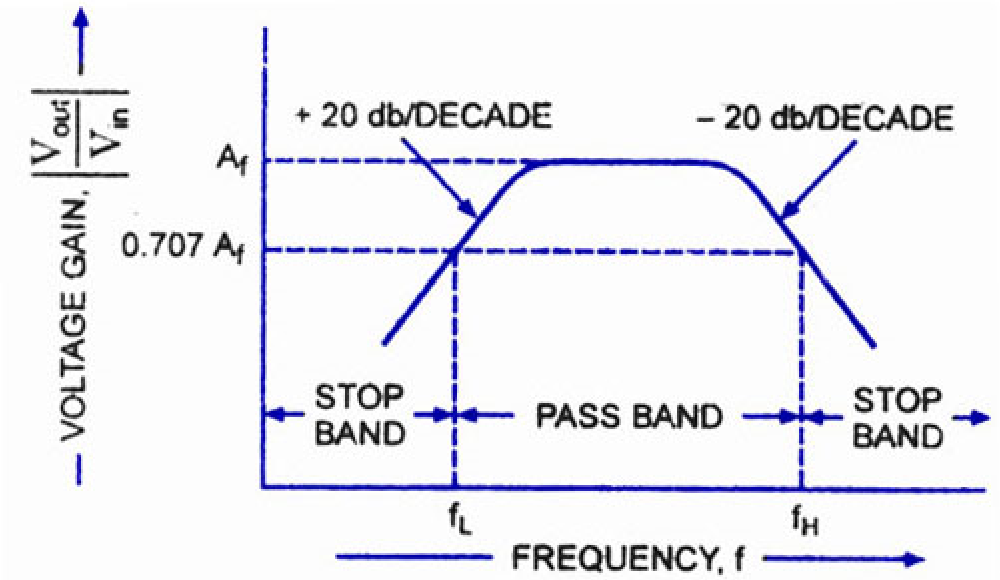

Figure 8: Band Pass Filter Frequency Response

7.4 Band Reject

It reject the frequency of desire band and pass all the frequency. It is the inverse of a band-pass filter. A notch filter is a band-stop filter with a narrow stop band (high Q component).

Narrow notch filters (optical) are utilized as a part of Raman spectroscopy, live solid multiplication (public address system, or PA frameworks) and in instrument amplifiers (particularly amplifiers or preamplifiers for acoustic instruments, for example, acoustic guitar, mandolin, bass instrument enhancer, and so forth.) to decrease or anticipate sound feedback. Different names incorporate 'band limit filter', 'T-notch filter', 'band-elimination filter', and 'band-reject filter'.

7.5 All Pass

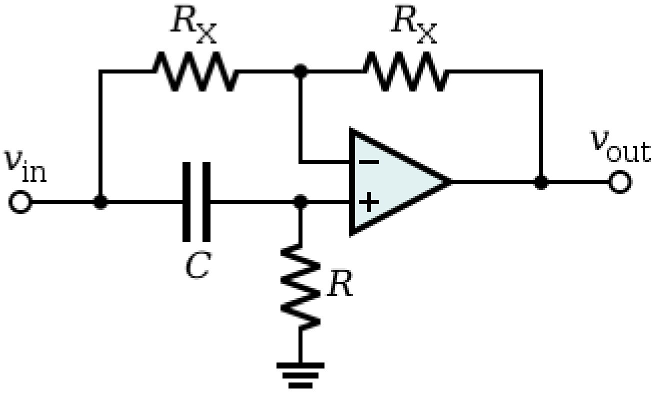

Figure 9: All Pass Filter

It pass all the frequency range. It may, changes the phase relationship between different frequencies. It does this by differing its phase shift as an element of frequency. For the most part, the filter is portrayed by the frequency at which the phase shift crosses 90 (i.e., when the input and output signals go into quadrature when there is a quarter wavelength of delay between them).

They are by and large used to make up for other undesired phase shifts that emerge in the system, or for blending with an unshifted form of the first to execute a notch comb filter.

They might likewise be utilized to change over a mixed phase filter into a minimum phase filter with an identical size reaction or an unstable filter into a steady filter with an equal magnitude response.

8 Types of Digital Filter

1. FIR (finite impulse response) filter
2. IIR (Infinite impulse response) filter

8.1 FIR Filter

In sign handling, a finite impulse response (FIR) filter is a filter whose impulse response (or response to any finite length information) is of finite span, in light of the fact that it settles to zero in finite time. This is rather than infinite impulse response (IIR) filters, which may have internal feedback and may keep on reacting indefinitely (normally rotting).

The impulse response (that is, the yield in response to a Kronecker delta data) of a Nth-request discrete time FIR filter endures precisely N + 1 specimens (from first nonzero component through last nonzero

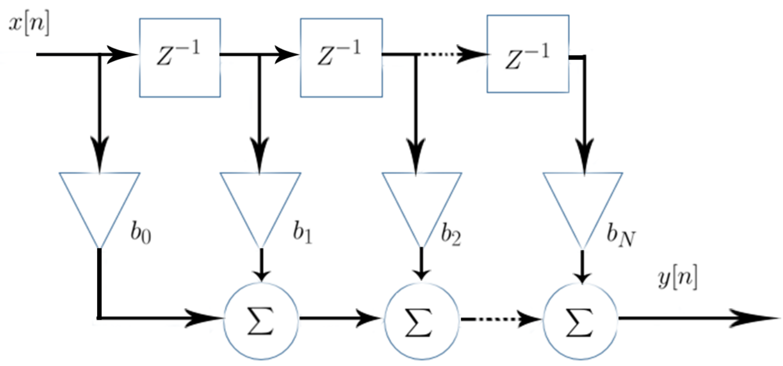

Figure 10: FIR Filter

component) before it then settles to zero.

FIR filters can be discrete-time or ceaseless time and advanced or simple.

Advantages

1. They are intrinsically steady
2. They can be plan to have a linear phase.
3. There is an incredible adaptability in forming their magnitude.
4. They are simple and advantageous to execute.

8.2 IIR Filter

Infinite impulse response (IIR) is a property applying to numerous linear time-invariant system. Regular cases of linear time-invariant system are most electronic and digital filter. System with this property are known as IIR system or IIR filters, and are recognized by having an impulse response which does not get to be precisely zero past a certain point, but rather proceeds uncertainly. This is rather than a limited impulse response in which the impulse response h(t) does get to be precisely zero now and again t ¿ T for some finite T, along these lines being of limited span.

9 Work Done

9.1 Procedure

In this project work I have collected the audio sample of ten different student. I have done the FFT (fast Fourier transform) of individual signal. After that I merge two or more audio signal i.e user can define it and done the FFT. We can use the instrumental signal for mixing. The power spectral density of the merge signal is drawn. After that the signal pass through a band pass FIR filter for filtering purpose seeing the power spectrum we can define the stop band and pass band of signal.

After that I have done the IFFT (Inverse Fourier transform) and retrieve the main audio signal.

9.2 Program and Simulation

For FFT of signal

$$x(t) = 5\sin(2\pi 10000t) + 10\sin(2\pi 20000t)$$

Program

```
% The following program takes input of sum of 2 sine waves of user defined
% amplitude and frequency. Finally it applies an positive FFT to the
% resulting data at a user defined sampling frequency and shows the spectrum
% Clearing Section
close all;
clear all;
clc;

% % Data Input
start_time = input('Enter the signal start time: ');
smp_frq = input('Enter the sampling frequency: ');
amp_1 = input('Enter the amplitude of the first sine wave: ');
freq_1 = input('Enter the frequency of the first sine wave in Hz: ');
amp_2 = input('Enter the amplitude of the second sine wave: ');
freq_2 = input('Enter the frequency of the second sine wave in Hz: ');
min_freq = gcd(freq_1,freq_2);
min_stop = 3/min_freq;
display('Minimum stop time for the data is ');
min_stop;
stop_time = input('Enter the signal stop time: ');

%% Preiodicity Checking
while(stop_time 〈 min_stop)
display('Minimum stop time for the data is ');
```

```
min_stop;
stop_time = input('Enter the signal stop time: ');
end;

    % % Data Interpretation
time_dom = [start_time:(1/smp_frq):stop_time];
sig_1 = amp_1sin(2πfreq_1 time_dom);
sig_2 = amp_2sin(2πfreq_2 time_dom);
data_any = sig_1 + sig_2;

    % % Fourier Tranfsform
[ YfreqD2,freqRng2 ] = positiveFFT(data_any,smp_frq);
stem(freqRng2,abs(YfreqD2));
grid on;
xlabel('Frequency in HZ—–>');
ylabel('Amplitude——>');
title('Frequency response of the given data');
yfft=YfreqD2_F*96000;
N=length(yfft);
freq_fft_psd = yfft(1:N);
psdx = (1/smp_frq)* abs(freq_fft_psd)^2;
psdx(1:end) = 2*psdx(1:end);
freq = 0:smp_frq/(2*N):smp_frq/2;
freq=freq(1:N);
subplot(2,1,1);
title('Periodogram Using FFT')
plot(freq,10*log10(psdx))S
grid on
xlabel('Frequency (Hz)')
ylabel('Power/Frequency (dB/Hz)')
subplot(2,1,2);
plot(freq,psdx);
xlabel('Frequency (Hz)')
```

```
ylabel('Power/Frequency (Watt/Hz)')
grid on;
```

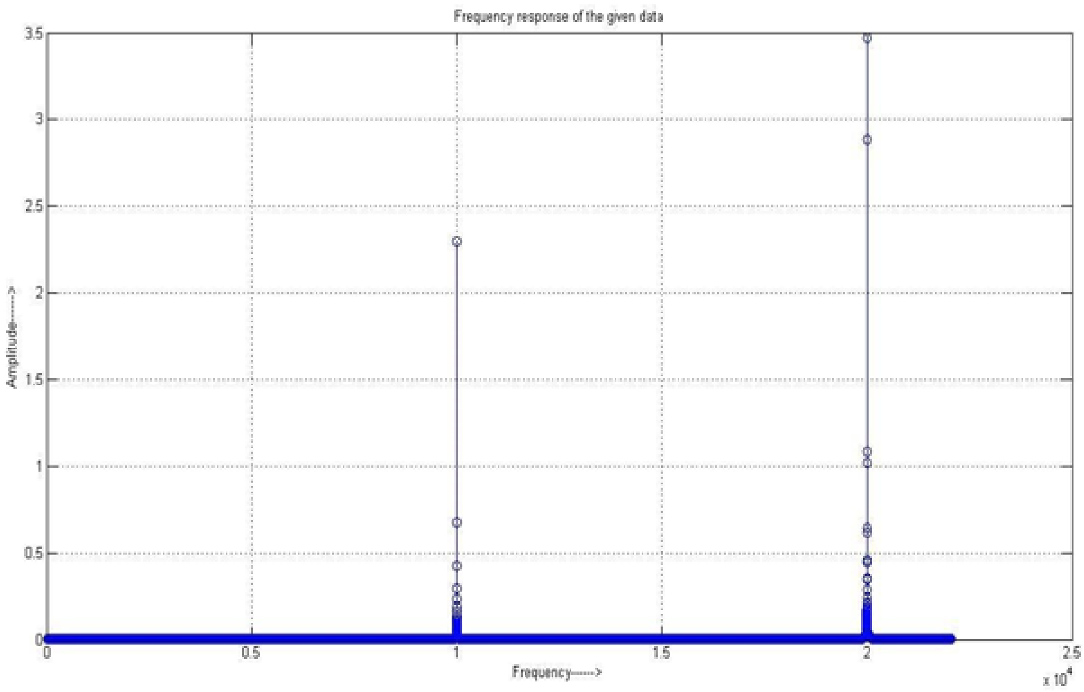

Figure 11: FFT OF signal x(t)=5 $\sin(2\pi 10000t) + 10\sin(2\pi 20000t)$

Program for FFT of Single and Multiple Audio Signal

```
% The following program takes input of sum of 2 sine waves of user defined
% amplitude and frequency. Finally it applies an positive FFT to the
% resulting data at a user defined sampling frequency and shows the spectrum
%% Clearing Section
close all;
clear all;
clc;

%% Data Input
N1 = input('Please enter number of sounds you would like to process: ');
YfreqD2_F = 0;
for i = 1:N1 start_time = 1;
strctaudio=uiimport();
y = strctaudio.data;
```

```
smp_frq = strctaudio.fs;
stop_time = 2*smp_frq;
data_any = y(1:stop_time);
if size(data_any,2)<=2
data_any=data_any';
end;
% data_any = y;

    %% Fourier Tranfsform
[YfreqD2,freqRng2] = positiveFFT_17k(data_any,44100);
YfreqD2_F = YfreqD2_F + YfreqD2;
end;
stem(freqRng2,abs(YfreqD2_F));
grid on;
xlabel('Frequency——>');
ylabel('Amplitude——>');
title('Frequency response of the given data');
```

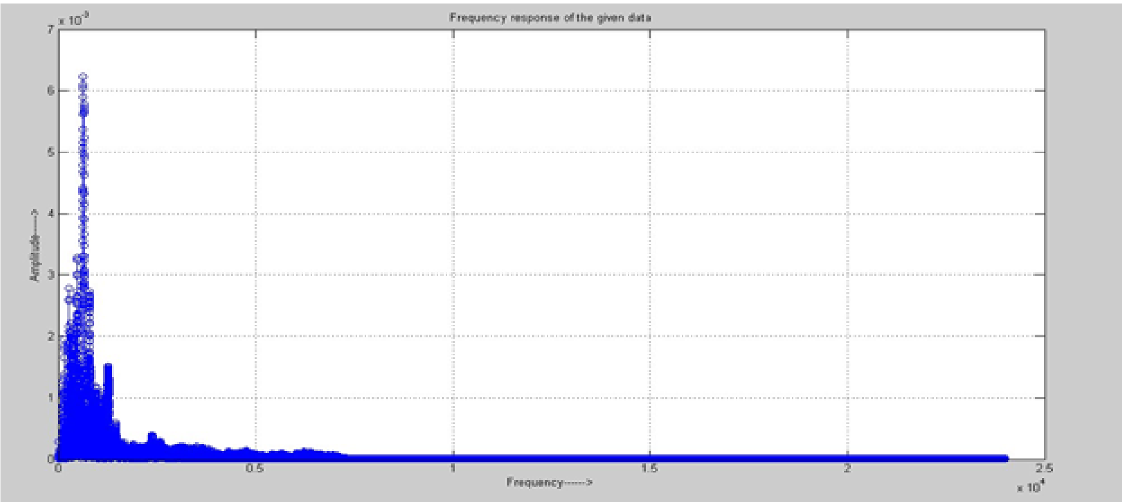

Figure 12: FFT of Single audio signal

```
    %The Signal is Passed through a High Pass Filter of Cutoff Frequency 5KHz.
%% Clearing Section
close all;
clear all;
```

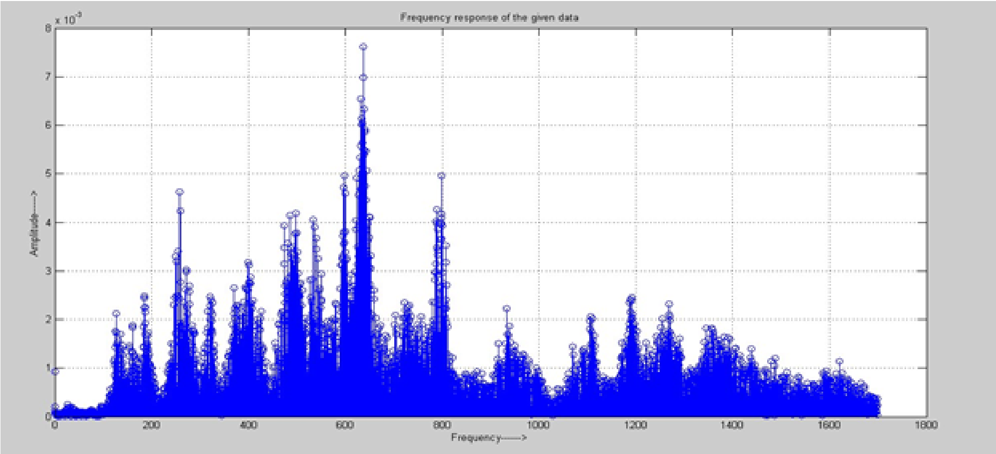

Figure 13: FFT of Multiple audio signal

```
clc;

    %% Data Input
start_time = 1;
strctaudio=uiimport();
y = strctaudio.data;
smp_frq = strctaudio.fs;
stop_time = 2*smp_frq;
% data_any = y(1:stop_time);
data_any = y;

    %% Fourier Tranfsform
[YfreqD2,freqRng2] = positiveFFT(data_any,smp_frq);
stem(freqRng2,abs(YfreqD2));
grid on;
xlabel('Frequency in HZ------>');
ylabel('Amplitude------>');
title('Frequency response of the given data bf');

    %% High Pass Filter
data_any_HPF = filter(FIR_Blackman_HPF_500(smp_frq),data_any);
% data_any_HPF = filter(IIR_HPF_500(smp_frq),data_any);
[YfreqD2_2,freqRng2_2] = positiveFFT(data_any_HPF,smp_frq);
```

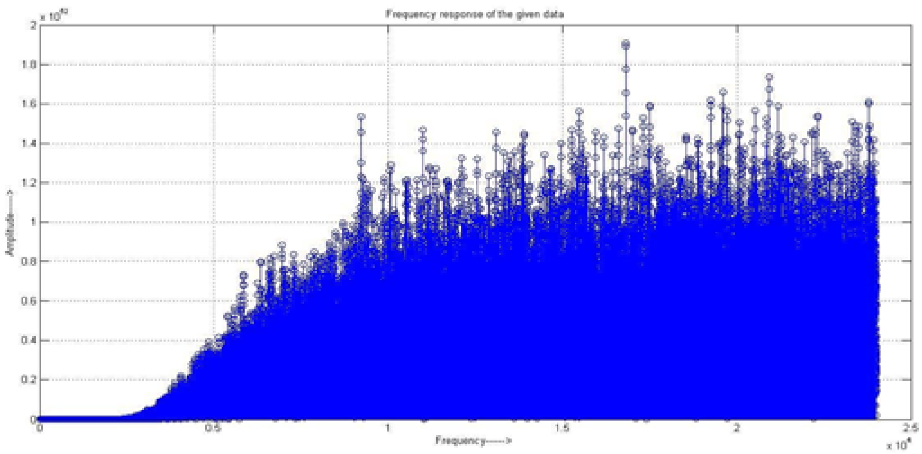

Figure 14: Audio spectrum of signal after filtering

figure;

stem(freqRng_2,abs(YfreqD2_2));

grid on;

xlabel('Frequency in HZ——>');

ylabel('Amplitude——>');

title('Frequency response of the given data af');

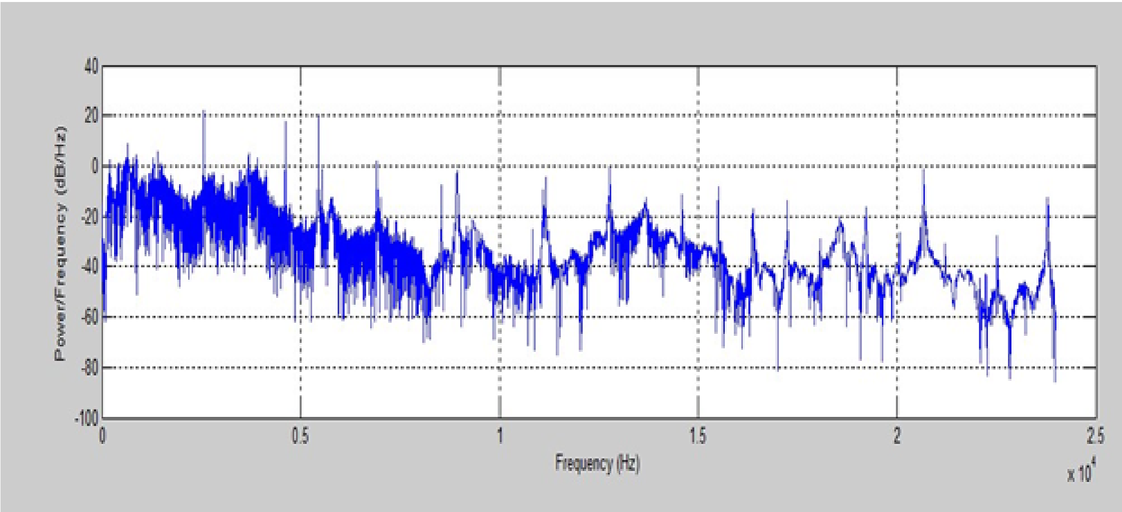

Figure 15: Power spectrum density of input signal

%Power Spectral density in psdx

figure;

yfft=YfreqD2_F*96000;

N=length(yfft);

```
freq_fft_psd = yfft(1:N);
psdx = (1/smp_frq)* abs(freq_fft_psd)^2;
psdx(1:end) = 2*psdx(1:end);
freq = 0:smp_frq/(2*N):smp_frq/2;
freq=freq(1:N);
subplot(2,1,1);
title('Periodogram Using FFT')
plot(freq,10*log10(psdx))
grid on
xlabel('Frequency (Hz)')
ylabel('Power/Frequency (dB/Hz)')
subplot(2,1,2);
plot(freq,psdx);
xlabel('Frequency (Hz)')
ylabel('Power/Frequency (Watt/Hz)')
grid on;
```

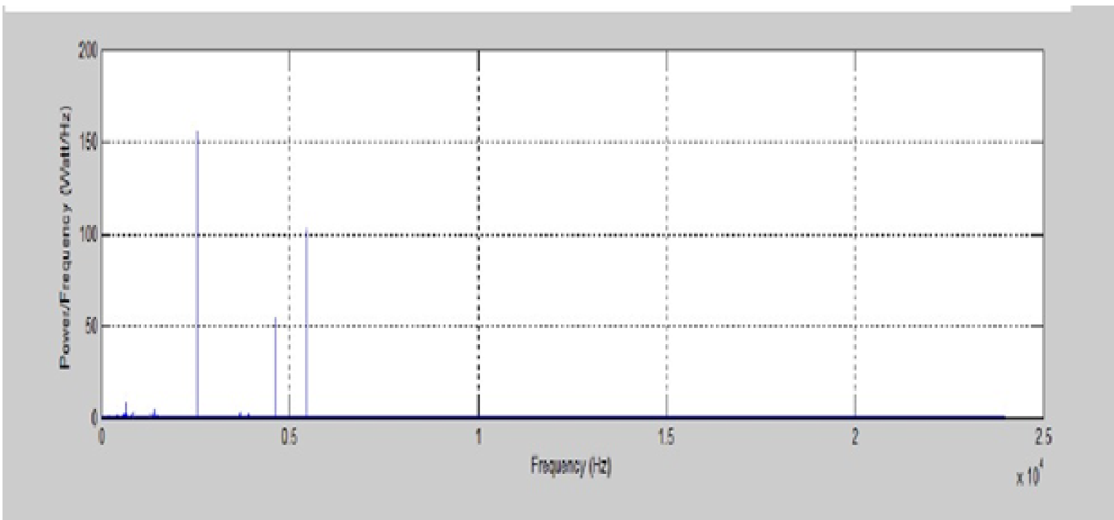

Figure 16: Power spectrum density of input signal

10 Conclusion

In this project work, I have taken the audio samples of ten different students and implemented the FFT (Fast Fourier Transform) of individual and multiple audio signals. I also have plotted the power spectrogram of the signals and mixed the audio signal with other signals like instrumental signals . As we obtain the power spectrum , the range of audio signal is known to us and we can use band pass filter to cut the region . By doing the IFFT (Inverse Fast Fourier Transform) I have retrieved the main signals. Initially, I studied the different blind beamforming systems and I have been able to implement them partially in this project work of mine. The major completed part of this project is the separation of audio signal component from mixture of acoustic audio signal.

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