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# Acoustic Noise Reduction in Speech Signals Using Microphone Array Beamforming Algorithms DSB, GSC and RGSC

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**ABSTRACT**: A microphone array consists of multiple microphones placed at different spatial locations. The multiple audio signals can be manipulated to enhance the desired signal and attenuate unwanted signals from particular directions. In this way, microphone arrays provide a means of enhancing a desired signal in the presence of corrupting noise sources. Microphone arrays have great potential in practical applications of speech processing, due to their ability to provide both noise robustness and hands-free signal acquisition by applying Beamforming algorithms. We are using three beamforming algorithms namely ***Delay and Sum Beamformer (DSB), Griffiths Jim Beamformer (GSC) and Robust Generalized Side lobe Canceller (RGSC)***. We simulated and compared the performance of these algorithms using MATLAB.

**KEYWORDS**: Microphone Array, DSB, GSC, Robust GSC, SNR.

## I. INTRODUCTION

Microphone is a device used to record Audio signals. Multiple Microphones together is a Microphone Array. When recording audio, it is important to eliminate all unwanted noise. Noise present due to the uncontrollable nature of a recording environment can be problematic to reduce as it consists of interfering sources and is statistically non-stationary. To reduce the noise and interference we apply Beamforming Algorithms.

Beamforming is achieved by filtering the microphone signals and combining the outputs to extract (by constructive combining) the desired signal and reject (by destructive combining) the interfering signals according to their spatial locations. Beamformers are classified as Fixed Beamformer and Adaptive Beamformer.

This paper aims to study the Delay and Sum Beamformer (DSB), Griffiths Jim Beamformer (GSC) and Robust Generalized Side lobe Canceller (RGSC).

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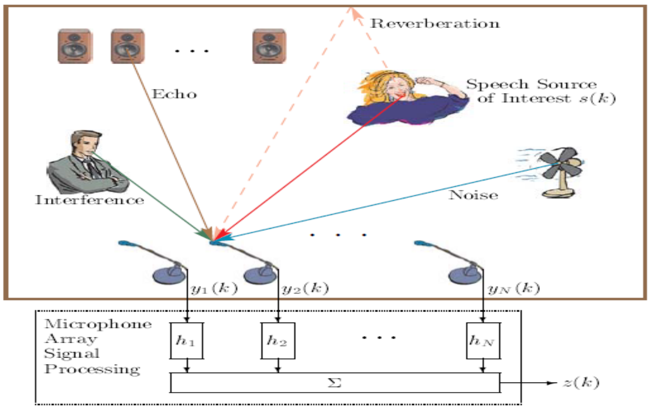


## II. RELATED WORK

In party environment there are multiple number of speakers, we want to listen a particular person audio signal, microphone is a device which is used to record the audio signal. A number of microphones are spatially distributed over the room. The output of the microphone array has target signal with noise and interference signals. We applied beamforming algorithms to the array output signal.The beamformer will enhance the desired signal and attenuates the unwanted signal. The output of the beamformer has high SNR compares to array output.

## III. PROPOSED ALGORITHM

### A. Design Considerations:

- Number of Microphones - 16 (Randomly placed)
- Source file - cocktail.wav
- Positions of mic and source - micpos.dat, srcpos.dat
- Wav file duration - 25 sec
- Sampling Frequency - 11025Hz

### B. Description of the Proposed Algorithms:

Aim of the proposed algorithms is to maximize the signal to noise ratio by minimizing the interference and noise signals in recorded audio signal.

**Delay and Sum Beamformer:**

In delay-and-sum beamforming, delays are inserted after each microphone to compensate for the arrival time differences of the speech signal to each to each microphone (Figure-1). The time aligned signals at the outputs of the delays are then summed together. This has the effect of reinforcing the desired speech signal while the unwanted off-axis noise signals are combined in a more unpredictable fashion. The signal-to-noise ratio (SNR) of the total signal is greater than (or at worst, equal to) that of any individual microphones signal. This system makes the array pattern more sensitive to sources from a particular desired direction.

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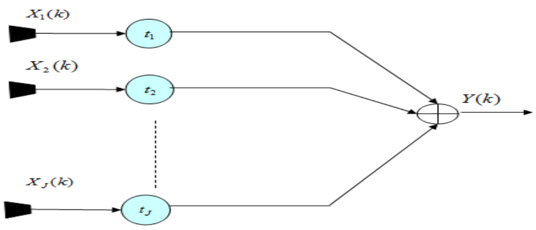


**Figure-1:** Block Diagram of Delay and Sum Beamformer

The Delay and Sum Beamformer output is given by

$$y(t) = 1/N \sum_{n=1}^{N} x_n(t-\tau_n) \qquad (1)$$

Where $\tau_n$is the delay samples, since the delay in between the microphones having a non-integer value of time samples, delay is done by phase change in Fourier domain instead of time domain [1].
The interspacing between microphones (d) is given by

$$d = c/2f_{max} \qquad (2)$$

**Griffiths-Jim Beamformer(Generalized Side Lobe Canceller):**

Generalized side lobe canceller is a flexible structure, since of its fixed and adaptive blocks are separated and individually manipulated. The Figure-2 shows the Generalized Side lobe Canceller in which it consists delay and sum Beamformer and blocking matrix blocks along with adaptive block [2].Adaptive part is simply group of filter that minimizes the power of the output, whereas blocking matrix (BM) used for minimize the noise power. Blocking matrix should be having (N-1) rows which are linearly independent microphones, the sum of the rows is zero and the rows are linearly independent. Thus the dimensions of BM must be (N-1) or less than that [3].

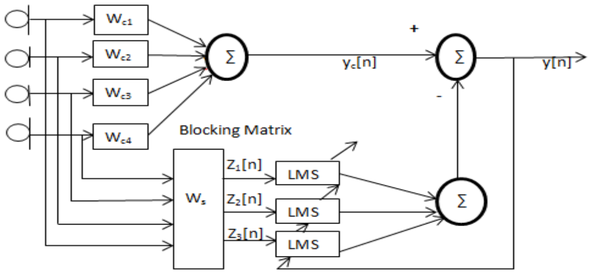


**Figure-2**:Block Diagram of Griffiths Jim Beamformer

Standard Griffiths-Jim Blocking Matrix (BM) is defined as

$$Ws = \left\{ \begin{matrix} 1 & -1 & 0 & 0 \\ 0 & 1 & -1 & 0 \\ 0 & 0 & 1 & -1 \end{matrix} \right\}$$

The output of BM is calculated as the matrix product of BM and the matrix of current input data.

$$z[n] = W_s.x[n] \qquad (1)$$

While in the adaptive section weight updating is done by using LMS algorithm and reference signal as y[n].

$$w_k [n+1] = w[n] + \mu y^*[n] z_k [n] \qquad (2)$$

Where "*" represents the conjugate value and "µ" is the step size. The final output for the GSC is given as

$$y[n] = y_c[n] - \sum_{k=1}^{N-1} w_k[n] .z_{k[n]} \qquad (3)$$

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Where $w_k[n]$ k is the $k^{th}$ column of the tap weight matrix $W_s$ and z[*n*] *k* is the $k^{th}$ blocking matrix output and these twomatriceshas same length [4].

**Robust Generalized Side lobe Canceller:**

In real world applications we can't estimate microphone characteristics perfectly. In such cases desired signal may leak into blocking matrix results in attenuation of signal output from fixed beamformer because some of its output is deleted because of desire signal presence in blocking matrix so here we use adaptive blocking matrix. This type of generalized beamformer is known as robust generalized side lobe canceller (RGSC) andFigure-3 shows the Robust GSC.

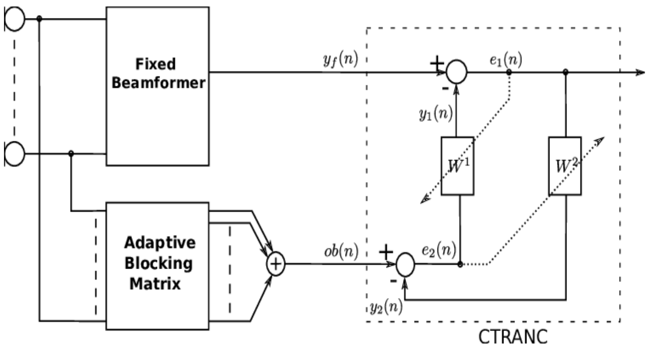


**Figure-3:** Block diagram of Robust Generalized Side Lobe Canceller

## IV. SIMULATION RESULTS

We are considering the Cocktail party environment. The cocktail party environment consists of four speakers in the room and 16 number of microphones which are distributed randomly in that room. Speaker-1 will be the target and other three speakers are interferers. The First speaker is talking about hockey. The second speaker is primary interferer he even closer to the Microphone closest to First speaker and talking louder.

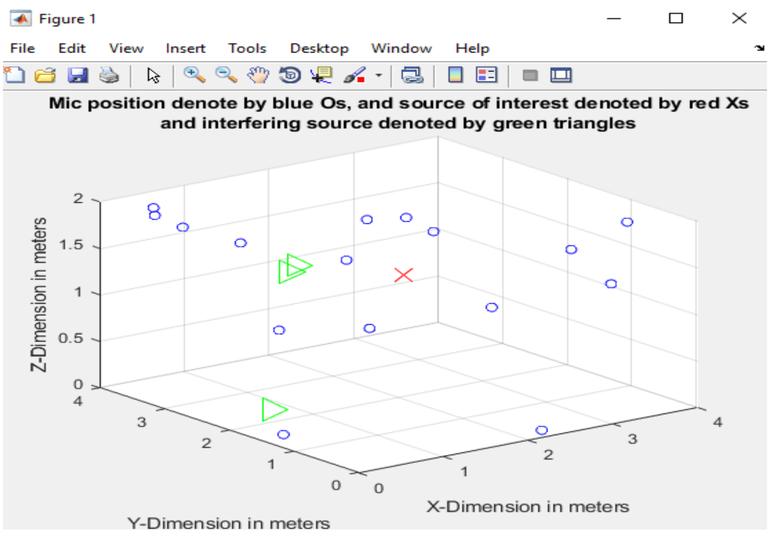


**Figure-1** Microphone, Source and Interfering source positions.

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It can be seen from Figure-1.It is a 3D plot that shows the Microphone positions in Blue( O's), Source position is Red(X) and Interfering source positions in Green(Triangles).We are taking the closest microphones near the target person. The output signal of the microphone array is applied to the beamforming algorithms. We are using three beamforming algorithms Delay and Sum Beamformer(DSB), Generalized Side lobe Canceller(GSC) and Robust Generalized Side lobe Canceller(RGSC).

**SIGNAL AT CLOSEST MIC:**

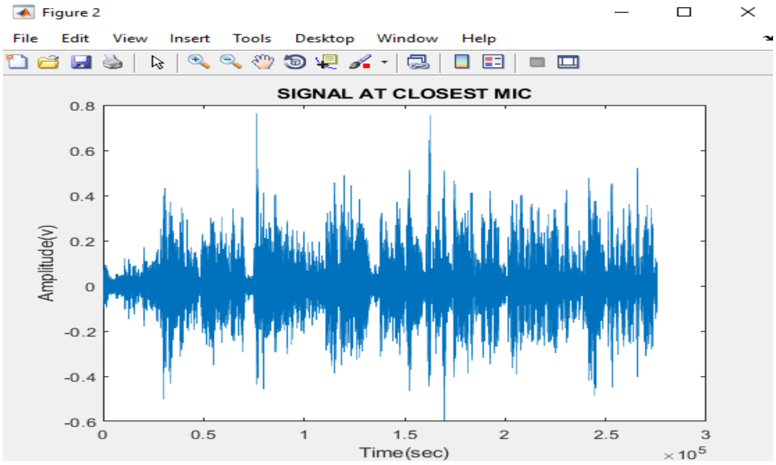


**Figure-2** Signal at Closest Mic

We are considering the closest Mic to the First Speaker who is talking about hockey. Above Figure-2 shows the output of the Signal at closest mic it consists of desired signal, noise and interference signal.

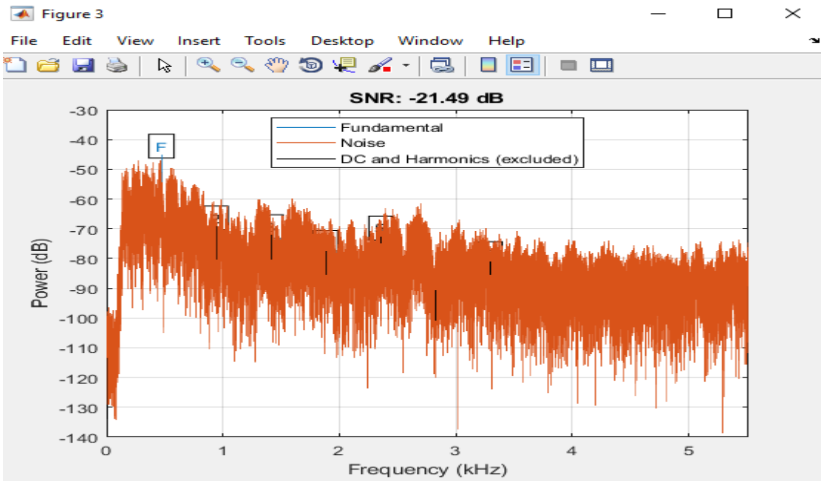


**Figure-3** SNR of Signal at Closest Mic

We have calculated the Signal to Noise ratio of Signal at closest mic. We can observe from the above Figure-3 that the Noise power is more than the signal power therefore the SNR value is very less. Fundamental is the desired signal.

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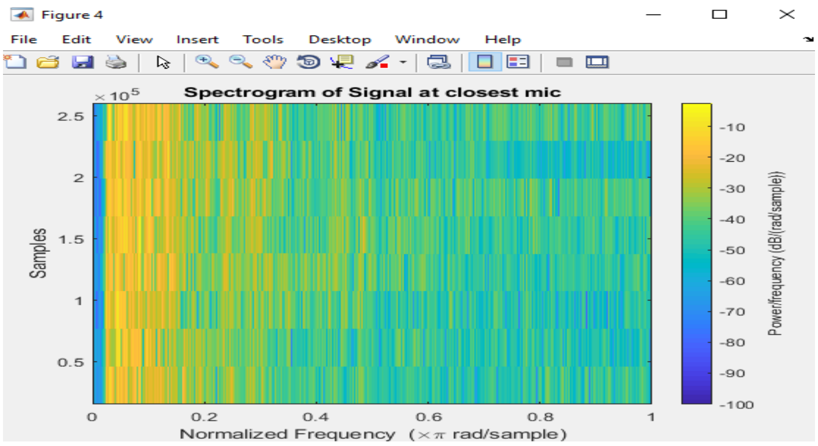


**Figure-4** Spectrogram of Signal at Closest Mic

Above Figure-4 shows the Spectrogram of Signal at closest mic. Spectrogram is a visual representation of the spectrum of frequencies of a signal as it varies with samples.

**DELAY AND SUM BEAMFORMER:**

In Delay and Sum Beamformer, delays are applied to 16 microphones and the outputs of the delay are summed together. By adding the signals the desired signal strength will be increased and Noise and interference signal strength decreased. So the voice clarity is increased than the signal at closest mic.

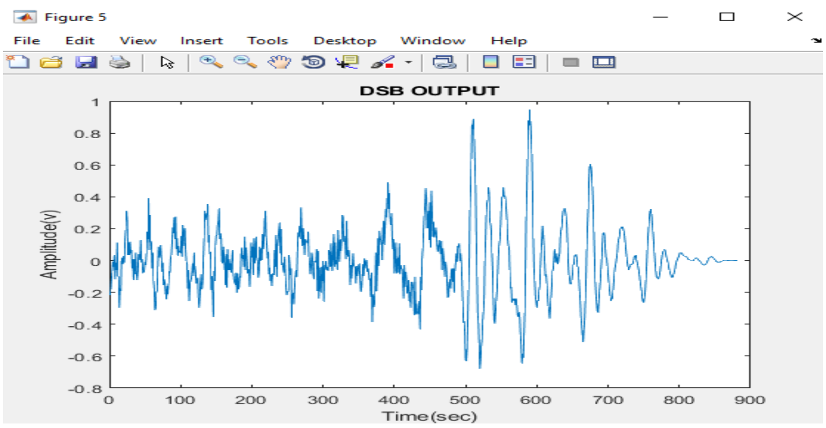


**Figure-5** DSB output Signal

Above Figure-5 shows the output of Delay and Sum beamformer. We can clearly observe that the DSB enhances the desired signal and attenuates the noise and interference signals.

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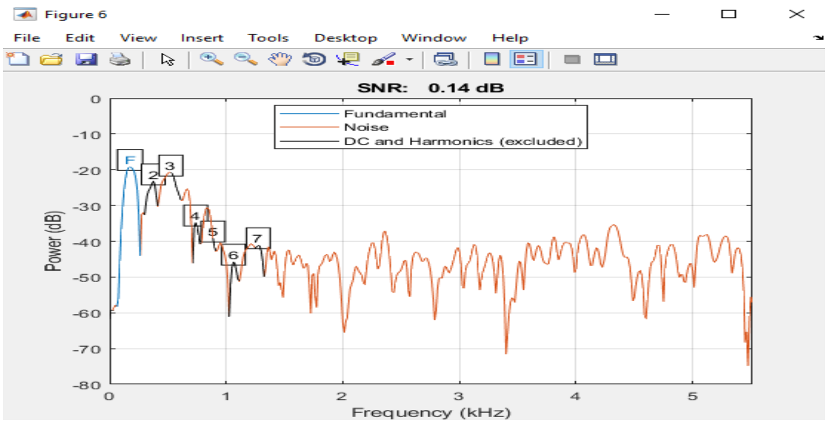


**Figure-6** SNR of DSB Output Signal

We have calculated the SNR value of DSB Signal, from above Figure-6 we can observe that the Fundamental signal (Desired signal) power is increased when compared to the Signal at closest mic, so that the SNR value also increased.

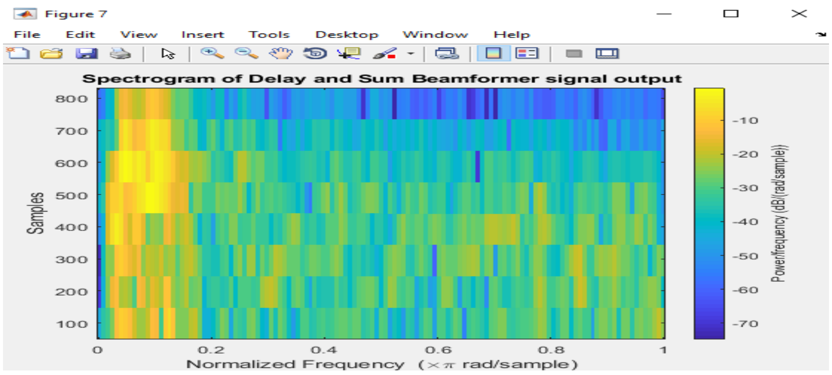


**Figure-7** Spectrogram of DSB Output Signal

Above Figure-7 shows the Spectrogram of DSB signal. Spectrogram is a visual representation of the spectrum of frequencies of a signal as it varies with samples.

**GENERALIZED SIDELOBE CANCELLER (GSC):**

Generalized side lobe canceller consists of two blocks fixed and adaptive blocks. The output of fixed block is target and interference signal and the output of the adaptive block is interference signal. The adaptive block output is subtracted from the output of the fixed block. The target signal may leak into the blocking matrix so the output of the adaptive block consists of target signal and interference signal so the target signal may also get cancelled.

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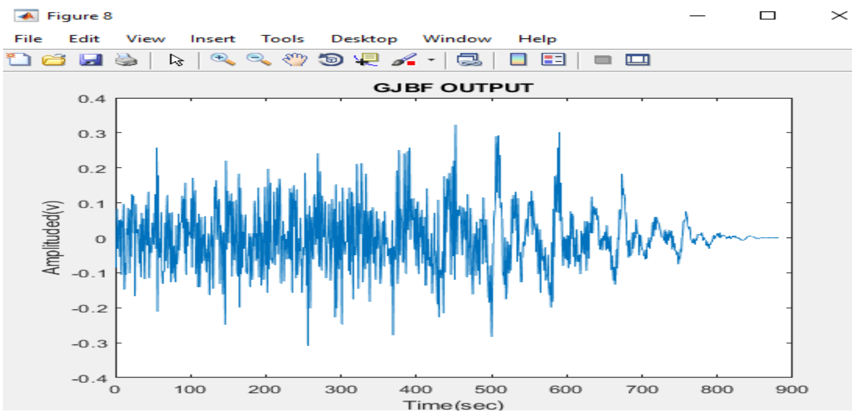


**Figure-8** GSC output signal

Above Figure-8 shows the output of GSC beamformer. We can observe that GSC has high noise power compared to DSB.

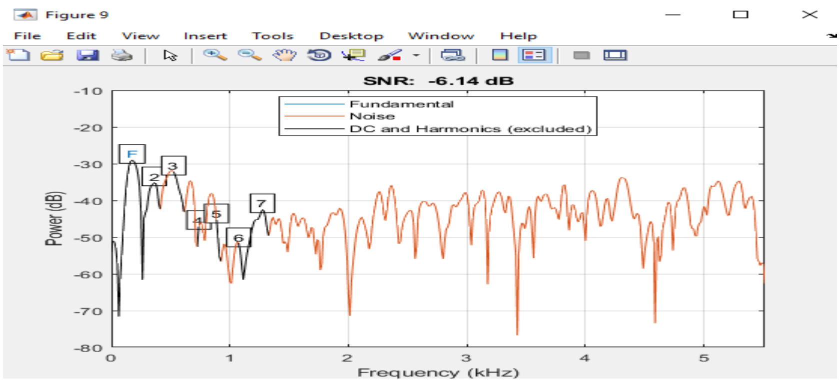


**Figure-9** SNR of GSC output signal

We have calculated the SNR value of GSC Signal, from above figure-9 we can observe that the Fundamental signal (Desired signal) power is decreased when compared to the DSB, so that the SNR value is also decreased.

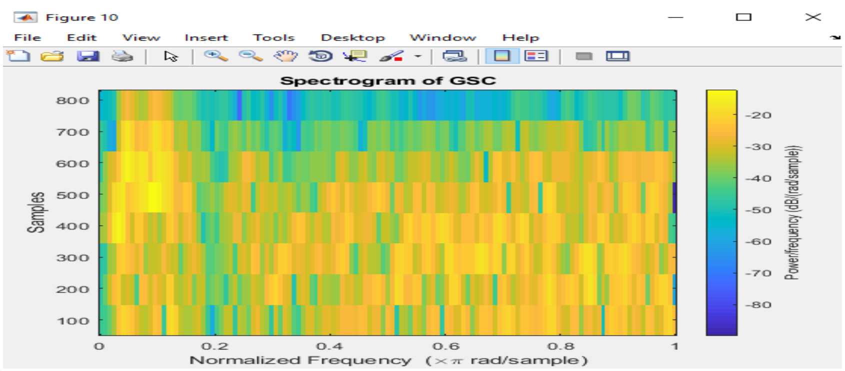


**Figure-10** Spectrogram of GSC

 

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Above Figure-10 shows the Spectrogram of GSC signal. Spectrogram is a visual representation of the spectrum of frequencies of a signal as it varies with samples.

**ROBUST GENERALISED SIDELOBE CANCELLER:**

Due to the drawback of GSC we have used Robust GSC. In this we use adaptive blocking matrix in order to avoid the leakage of target signal.

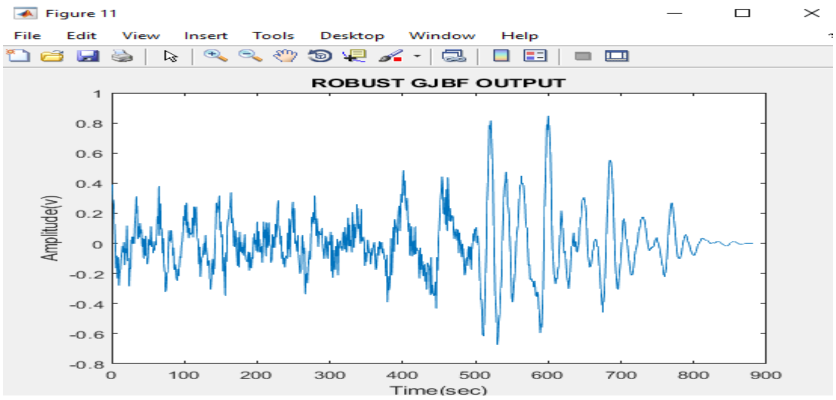


**Figure-11** Robust GSC output signal

Above Figure-11 shows the output of Robust GSC. When compared to DSB and GSC beamformers Robust GSC has more signal power.

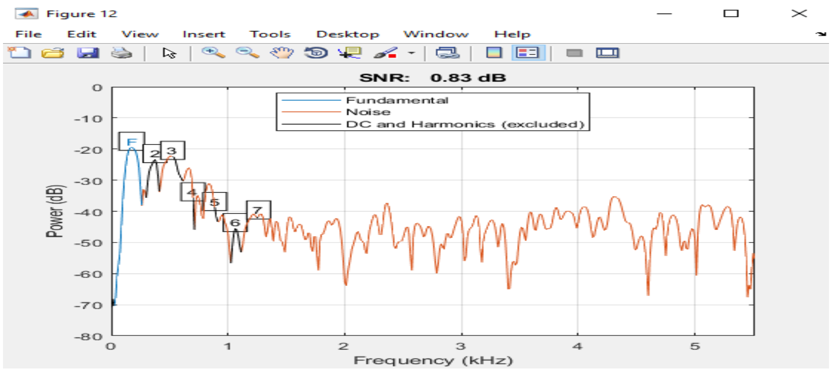


**Figure-12** SNR of Robust GSC output signal

We have calculated the SNR value of Robust GSC Signal, from above figure-12 we can observe that the Fundamental signal (Desired signal) power is increased when compared to the GSC, so that the SNR value is also increased.

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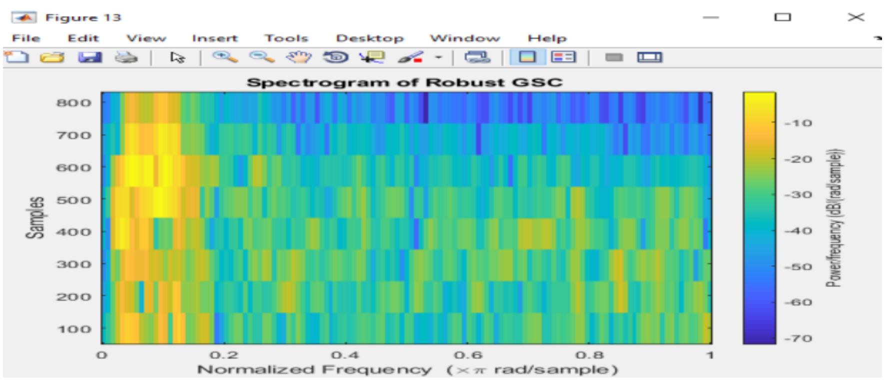
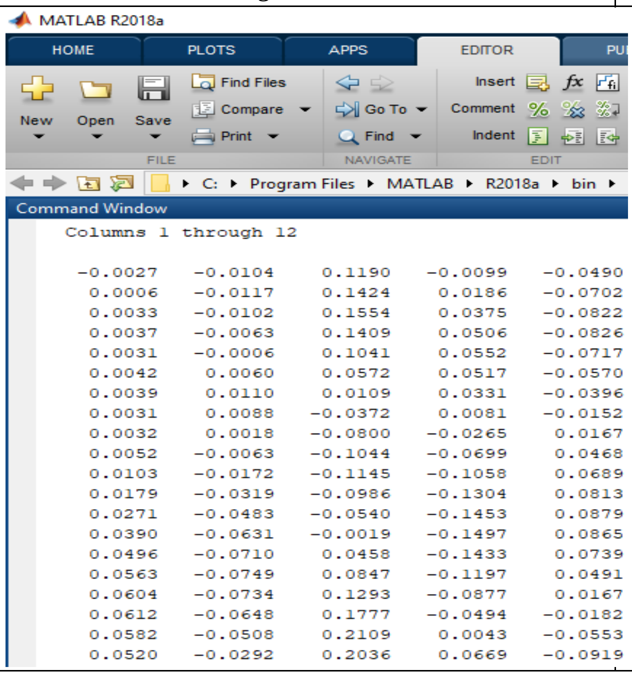



**Figure-13** Spectrogram of Robust GSC

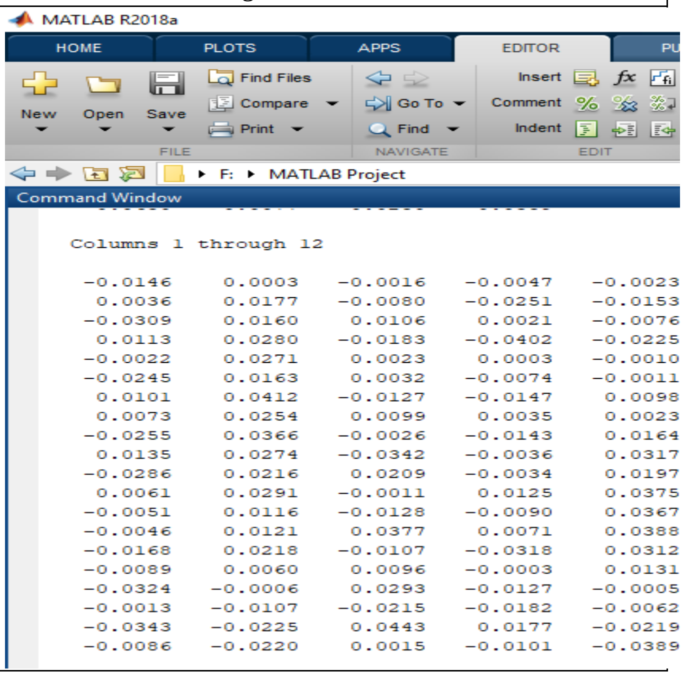
Above Figure-13 shows Spectrogram of Robust GSC signal. Spectrogram is a visual representation of the spectrum of frequencies of a signal as it varies with samples.

| Weights of GSC | Weights of Robust GSC |
|---|---|



Columns 1 through 12

| | | | | |
|---|---|---|---|---|
| -0.0027 | -0.0104 | 0.1190 | -0.0099 | -0.0490 |
| 0.0006 | -0.0117 | 0.1424 | 0.0186 | -0.0702 |
| 0.0033 | -0.0102 | 0.1554 | 0.0375 | -0.0822 |
| 0.0037 | -0.0063 | 0.1409 | 0.0506 | -0.0826 |
| 0.0031 | -0.0006 | 0.1041 | 0.0552 | -0.0717 |
| 0.0042 | 0.0060 | 0.0572 | 0.0517 | -0.0570 |
| 0.0039 | 0.0110 | 0.0109 | 0.0331 | -0.0396 |
| 0.0031 | 0.0088 | -0.0372 | 0.0081 | -0.0152 |
| 0.0032 | 0.0018 | -0.0800 | -0.0265 | 0.0167 |
| 0.0052 | -0.0063 | -0.1044 | -0.0699 | 0.0468 |
| 0.0103 | -0.0172 | -0.1145 | -0.1058 | 0.0689 |
| 0.0179 | -0.0319 | -0.0986 | -0.1304 | 0.0813 |
| 0.0271 | -0.0483 | -0.0540 | -0.1453 | 0.0879 |
| 0.0390 | -0.0631 | -0.0019 | -0.1497 | 0.0865 |
| 0.0496 | -0.0710 | 0.0458 | -0.1433 | 0.0739 |
| 0.0563 | -0.0749 | 0.0847 | -0.1197 | 0.0491 |
| 0.0604 | -0.0734 | 0.1293 | -0.0877 | 0.0167 |
| 0.0612 | -0.0648 | 0.1777 | -0.0494 | -0.0182 |
| 0.0582 | -0.0508 | 0.2109 | 0.0043 | -0.0553 |
| 0.0520 | -0.0292 | 0.2036 | 0.0669 | -0.0919 |



Columns 1 through 12

| | | | | |
|---|---|---|---|---|
| -0.0146 | 0.0003 | -0.0016 | -0.0047 | -0.0023 |
| 0.0036 | 0.0177 | -0.0080 | -0.0251 | -0.0153 |
| -0.0309 | 0.0160 | 0.0106 | 0.0021 | -0.0076 |
| 0.0113 | 0.0280 | -0.0183 | -0.0402 | -0.0225 |
| -0.0022 | 0.0271 | 0.0023 | 0.0003 | -0.0010 |
| -0.0245 | 0.0163 | 0.0032 | -0.0074 | -0.0011 |
| 0.0101 | 0.0412 | -0.0127 | -0.0147 | 0.0098 |
| 0.0073 | 0.0254 | 0.0099 | 0.0035 | 0.0023 |
| -0.0255 | 0.0366 | -0.0026 | -0.0143 | 0.0164 |
| 0.0135 | 0.0274 | -0.0342 | -0.0036 | 0.0317 |
| -0.0286 | 0.0216 | 0.0209 | -0.0034 | 0.0197 |
| 0.0061 | 0.0291 | -0.0011 | 0.0125 | 0.0375 |
| -0.0051 | 0.0116 | -0.0128 | -0.0090 | 0.0367 |
| -0.0046 | 0.0121 | 0.0377 | 0.0071 | 0.0388 |
| -0.0168 | 0.0218 | -0.0107 | -0.0318 | 0.0312 |
| -0.0089 | 0.0060 | 0.0096 | -0.0003 | 0.0131 |
| -0.0324 | -0.0006 | 0.0293 | -0.0127 | -0.0005 |
| -0.0013 | -0.0107 | -0.0215 | -0.0182 | -0.0062 |
| -0.0343 | -0.0225 | 0.0443 | 0.0177 | -0.0219 |
| -0.0086 | -0.0220 | 0.0015 | -0.0101 | -0.0389 |

**Figure-14** Weights of Generalized Side lobe canceller And Robust Generalized Side lobe canceller

Above Figure-14 shows the adaptive beamforming weights for generalized side lobe canceller and Robust Generalized side lobe canceller. Column represents the microphone array track.

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**Table-1**: SNR values for corresponding Beamforming Algorithm.

| SIGNAL AT MIC SNR(dB) | DSB SNR(dB) | GSC SNR(dB) | ROBUST GSC SNR(dB) |
|---|---|---|---|
| -21.4905 | 0.1385 | -6.1448 | 0.8280 |

Above Table-1 shows the values of SNR for all three beamforming algorithms. Robust Generalized Side Lobe Canceller has high SNR value compared to other two beamforming algorithms.

**Table2**. Comparison SNR vs. Order for Adaptive Beamforming Algorithms

| ORDER | GSC SNR(dB) | ROBUST GSC SNR (dB) |
|---|---|---|
| 5 | -6.8697 | 0.4005 |
| 10 | -6.0776 | 0.9418 |
| 15 | -6.1823 | 0.8144 |
| 20 | -6.1314 | 0.8232 |
| 25 | -6.2123 | 0.9199 |
| 30 | -6.6871 | 1.0961 |

Above Table-2 shows the SNR values for GSC and Robust GSC Beamforming Algorithms for different order values.

## V. CONCLUSION

We have observed that there is a significant improvement of SNR of the signal while using Beamforming Algorithms in Noisy environments. We have listened to the three beamforming outputs and also calculated the SNR values.Robust Generalized Side Lobe Canceller has high SNR value and has more voice clarity than Delay and Sum Beamformer and Generalized Side lobe canceller.

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