Adaptive Beamforming for Handsfree Telephony in Car

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Abstract

Adaptive beamforming system is studied to enhance speech acquisition for hands free telephony in car. A microphone array with an effective combination of spatial and temporal processing will lead to a very efficient solution compared to a single microphone. This process of beamforming has to be adapted for changing noise conditions. The adaptive beamforming is achieved using General Sidelobe Canceller (GSC) algorithm. In the GSC there are two parts, one is a fixed beamforming design that determines the response for the desired source and other is blocking matrix followed by multiple input canceller. The sidelobe canceller is evaluated using Normalized Least Mean Square (NLMS) algorithm. The system is simulated using MATLAB. The algorithm is tested using different interference signals including the speech signal of person other than the driver.

Introduction

Increased use of mobile telephones in cars has created a greater demand for hands-free in-car installations. The advantages of hands-free telephones are safety and convenience. In many countries and regions hand-held telephony in cars is prohibited by legislation. The car manufacturers also prohibit such use since it will interact with other electronic devices in the car such as air bags, navigation equipment etc. This means that a mobile telephone should be properly installed and an external antenna should be used. However, the main reason is safety and convenience. By installing the microphone far away from the talker, a number of disadvantages such as poor sound quality and acoustic feedback of the far-end side, are obvious. This means that filtering is required to obtain a similar sound quality as for hand held telephony. This filtering operation must suppress the loudspeaker, as well as background noise and room reverberation, without causing severe speech distortion. A microphone array with an effective combination of spatial and temporal processing will lead to a very efficient solution compared to a single microphone [4].

Adaptive Beamforming

Beamforming techniques can be used to enhance the speech signal[4]. Beamforming is the combination of radio signals from a set of small non-directional antennas to simulate a large directional antenna. The simulated antenna can be pointed electronically, although the antenna does not physically move. In communications, beamforming is used to point an antenna at the signal source to reduce interference and improve communication quality. In beamforming, both the amplitude and phase of each antenna element are controlled. Combined amplitude and phase control can be used to adjust side lobe levels and steer nulls better than can be achieved by phase control alone. The combined relative amplitude a_k and phase shift θ_k for each antenna is called a "complex weight" and is represented by a complex constant wk (for the k^{th} antenna).A beamformer for a radio transmitter applies the complex weight to the transmit signal (shifts the phase and sets the amplitude) for each element of the antenna array [5].

In a fixed beamformer , the weights may be chosen to give one central beam in desired direction. In the adapative beamforming, the complex weights are changed to steer the beam until maximum signal strength occurs in the direction of the signal source. some commonly used methods:

Minimum Mean-Square Error :The shape of the desired received signal waveform is known by the receiver. Complex weights are adjusted to minimize the mean-square error between the beamformer output and the expected signal waveform.

Maximum Signal-to-Interference Ratio: Where the receiver can estimate the strengths of the desired signal and of an interfering signal, weights are adjusted to maximize the ratio.

Minimum Variance: When the signal shape and source direction are both known, chose the weights to minimize the noise on the beamformer output.

Direction Finding

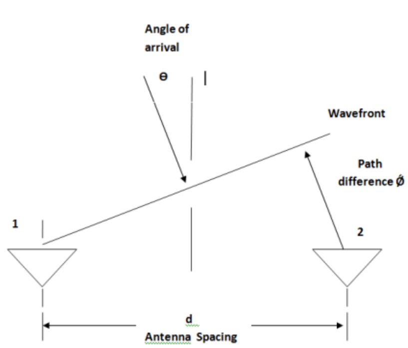

Fig .1 Wavefront arrival at the antenna arrays

A wave front from direction θ arrives at antenna 1 first. Then, after travelling an additional path distance Δl it arrives at antenna 2.

$\Delta l = d \sin\theta$

The pah difference results in a phase difference $\Delta\varphi$ between the signals from the two antennas: $\Delta\varphi = 2\pi\Delta l / \lambda$

A direction-finding system calculates the angle of arrival from the phase difference:

$$\Delta\varphi = 2\pi d \sin \theta / \lambda$$

A direction-finding system calculates the angle of arrival from the phase difference:

$$\theta = \sin^{-1} (\Delta\varphi \lambda / (2\pi d))$$

For a super-resolution result to be accurate, the arriving wave must be a direct signal from the source – a "plane wave" with a straight wavefront. Signal reflections (multipath) and interfering signals cause super- resolution systems to fail. A super-resolution system cannot operate if two or more signal sources share the same frequency, since the receiver's output phase no longer reflects the phase of an incoming plane wave.

Beamforming can be used for direction finding by rotating the central beam of an array to give maximum received signal strength. With this method, the angular resolution is limited by the beam width produced by the beamformer. Also, false measurements will occur if a side lobe is mistakenly steered to the signal source, instead of the array's central lobe. However, it is possible to measure the directions of multiple sources and to identify the directions of reflections with a beamforming system.

Linearly Constrained Adaptive Beamforming

The adaptive beamforming requires the sum of the array signals is minimized but with a constraint that the signal in the source direction should not be altered and all the interference signals in the different directions should be cancelled. The linearly constrained adaptive algorithm may be implemented using the structure shown in Fig.2. Time delay steering elements T1,T2,….Tm are used to point the array in the direction of interest. This implementation is called as the direct form. Each coefficient in the beamformer is updated by the adaptive processor, which computes new values using the algorithm.

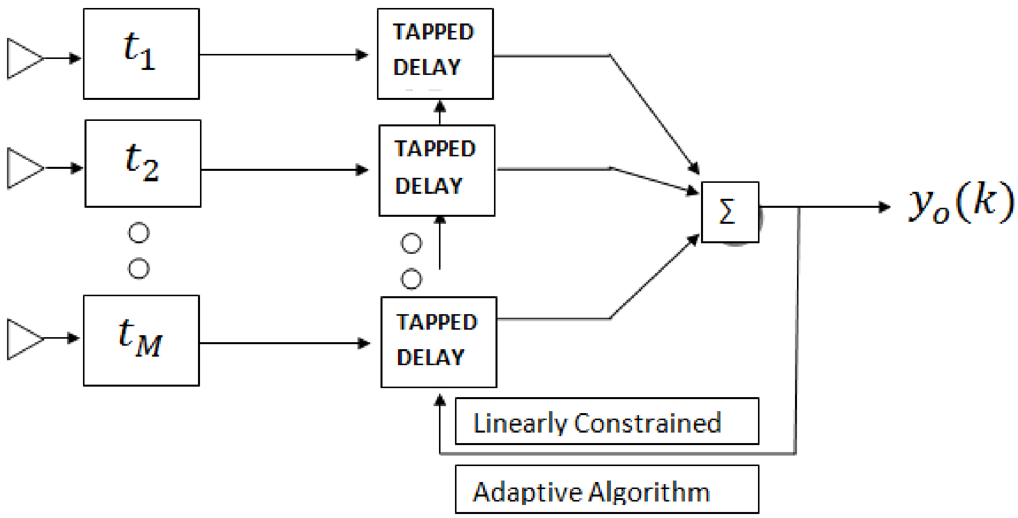

Fig .2 Direct form implementation of linearly constrained adaptive array processing algorithm

The observation vector x(n) and a desire response d(n), and the tap weight W so that

$$\mathrm{e}(n) = d(n) - w^T x(n)$$

e(n) can be minized in the mean square sense and the constraint is

$$c^T w = a$$

Where c is the column vector and a is the scalar

This equation can be solved by using Lagrange multiplier, According to method of the Lagrange multiplier.

$$\xi^c = E[e^2(n)] + \lambda(c^T w - a)$$

here λ is the Lagrange multiples. And solve the equation .

$$\nabla_w \xi^c = 0 \ \ and \ \ \frac{\partial \xi^c}{\partial \lambda} = 0$$

$$\xi^c = \xi_{min} + v^T R v + \lambda(c^T v - a')$$

Where v =w-w_0

Consider the two element narrow band beam former given by

$$x_0(n) = \alpha(n)e^{j\Phi 1} + \beta(n)e^{j(\Phi 2-\Phi 0)}$$

$$x_1(n) = \alpha(n)e^{j\Phi 1} + \beta(n)e^{j\Phi 2}$$

Where $\alpha(n)$ is the source signal and $\beta(n)$ is the noise signal .

The beam former tap weights, w_0 and w_1 are adjusted so that their output y(n) minimized mean square sense . Following constraint must be hold

$$w_0 + w_1 = 1 \ and \ w^H c = 1$$

Where c=$[1\ 1]^T$ and w=$[w_0^*\ w_{01}^*\ ]^T$

$$y_0(n) = (w_0^c)^H x(n) = \alpha(n)e^{j\Phi 1}$$

Finally output signal consists of source signal $\alpha(n)$ alone .

Often, constraints are placed on the adaptive beamformer so that the complex weights do not vary randomly in poor signal conditions. Some radio signals include “training sequences” so that an adaptive beamformer may quickly optimize its radiation pattern before the useful information is transmitted.

For two antenna elements spaced at 4 wavelengths, the following diagram shows the radiation pattern that a beamformer would produce. The main drawback of the beamforming approach - many side lobes unless many antenna elements are used - is apparent. A super-resolution system using the same two antennas could measure direction accurately, provided that the only an undistorted plane wave is arriving.

Fig. 3 Array of microphone set up in car .

Generalized Sidelobe Canceler (GSC)

The linear constrained algorithm can also be implemented using unconstrained adaptive algorithms by using GSC algorithm [7]. The general layout of GSC is shown in Figure 4. The four microphone outputs are time delayed steered to produce 4 signals which ideally have the desired signal in phase with each other. These 4 signals are then sent to the blocking matrix. The purpose of the blocking matrix is to block out the desired signal from the lower part of the GSC. We want to adaptively cancel out noise, therefore we only want noise to go into the adaptive filters FIR1, FIR2, and FIR3. In the implementation used in this thesis, the blocking matrix takes the following form:

$$\mathbf{B} = \begin{bmatrix} 1 & -1 & 0 & 0 \\ 0 & 1 & -1 & 0 \\ 0 & 0 & 1 & -1 \end{bmatrix}$$

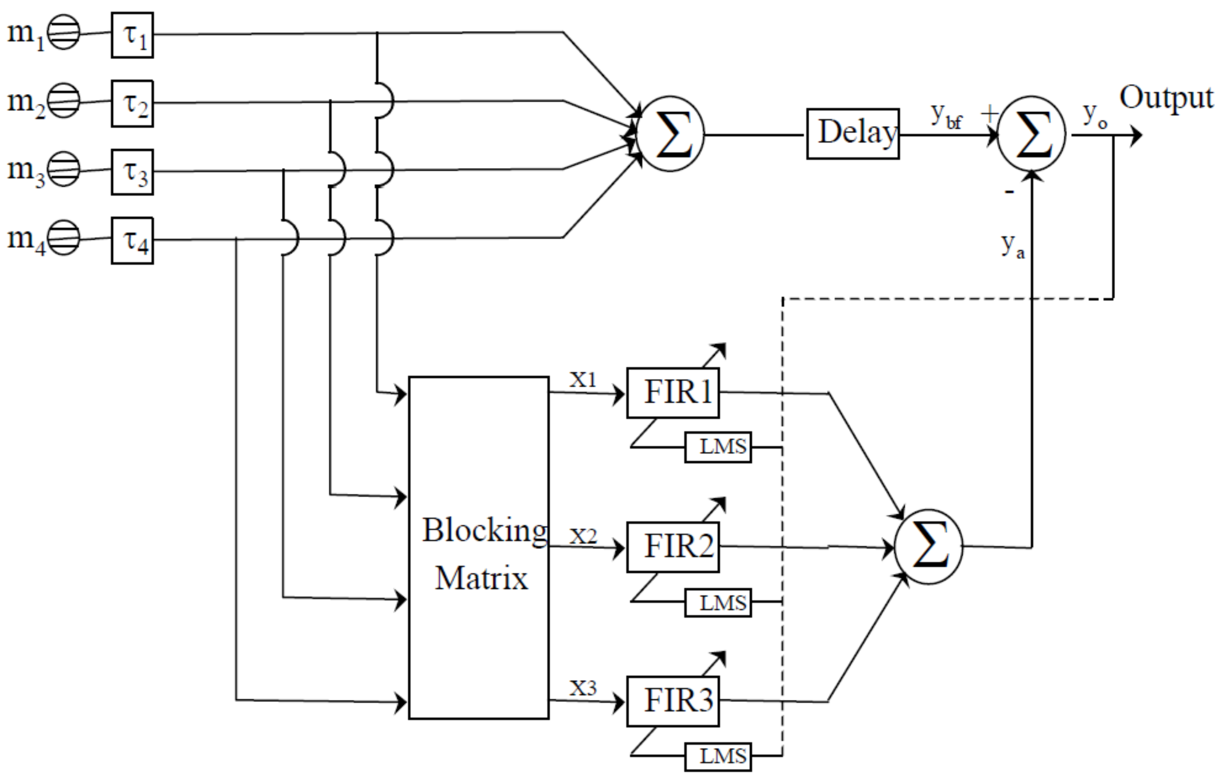

Fig .4 Generalized sidelobe canceler block diagram

This just means that the outputs of the blocking matrix are the difference between successive signal samples. If the inputs to the blocking matrix are perfectly in phase, then this form works fine. The top branch of the sidelobe canceller produces the beamformed signal. The bottom branch, after the blocking matrix, contains 3 filtered versions of the noise components in the signal in the top branch. Now we would like to combine these three signals in such a way as to best approximate the noise component in the upper (beamformed) signal, generating a linear combination of filtered noise components as the approximation. This can be done by minimizing the output of the GSC, which is the signal from the upper (beamformed) channel minus the signal from the lower (noise approximation) channel. Now we have to choose which method we want to use to adapt the FIR filter weights.

The ordinary LMS algorithm requires a user-specified step-size for filter Coefficient updating. We would like the step-size to be correlated with the power of the received signals, such that we are confident about the convergence of the algorithm. The received signals change in power over time, so that the NLMS algorithm is a better approach. Now we need to establish some notation. The vectors $\mathbf{a}_{1,k}$, $\mathbf{a}_{2,k}$, and $\mathbf{a}_{3,k}$ contain the filter coefficients of FIR1, FIR2, and FIR3, respectively, at time *k*.

These coefficient vectors can be combined to create an overall filter coefficient vector and the vectors $\mathbf{x}_{1,k}$, $\mathbf{x}_{2,k}$, and $\mathbf{x}_{3,k}$ contain the inputs into FIR1, FIR2, and FIR3, respectively, at time *k*. Each $\mathbf{x}_{i,k}$ satisfies the following equation:

$$a_k = \begin{bmatrix} a_{1,k} \\ a_{2,k} \\ a_{3,k} \end{bmatrix} \qquad X_{i,k} = \begin{bmatrix} X_{i,k-\phi} \\ X_{i,k-\phi+1} \\ \cdot \\ \cdot \\ X_{i,k-\phi+1} \end{bmatrix}$$

where ϕ is the filter order. The overall input vector is

$$X_k = \begin{bmatrix} X_{1,k} \\ X_{2,k} \\ X_{3,k} \end{bmatrix}$$

Now we can write the NLMS filter update equation at time *k*+1:

$$a_{k+1} = a_k + \frac{\mu}{||X_K||^2} X_k Y_{o,k}$$

where is the step size. The current output, *yo,k*, is

$Y_{ko,k} = Y_{bf,k} - Y_{a,k}$ where $Y_{a,k} = a_k^H X_k$

This algorithm requires 3(ϕ +1) multiplies, 3(ϕ +1)+1 adds, 3(ϕ +1) multiply-adds, and 3(ϕ +1) divisions. Fortunately, the updating for the filters can be done in parallel with 3 processors, thus reducing the computational complexity by approximately 3. This is one advantage of using the NLMS algorithm.

Finally, a delay is introduced in the beamformer section of the GSC. This delayoccurs before the adaptive component *ya*(*k*) is subtracted from the beamformer component *ybf*(*k*) to produce *yo*(*k*). This delay is corresponding to the target speech filter of the GSC.

Simulation

We have made the set up with 4 microphone aligned in straight line with 4 cm apart corresponding to λ/2 of sound wave having maximum speech frequency 4 KHz .We simulated speech signal with different Interference signals by adding the speech signal with differently delayed interference signals. Source signal is assumed to be arriving at o degree and interference signals at an angle 45 degree. We used the filter order of 14 as the results were good for order between 10 to 20.

Results

We tested this set up in lab with different noise condition**:**

1. We arranged the set up with 0 degree for source signal and interference from all other direction.

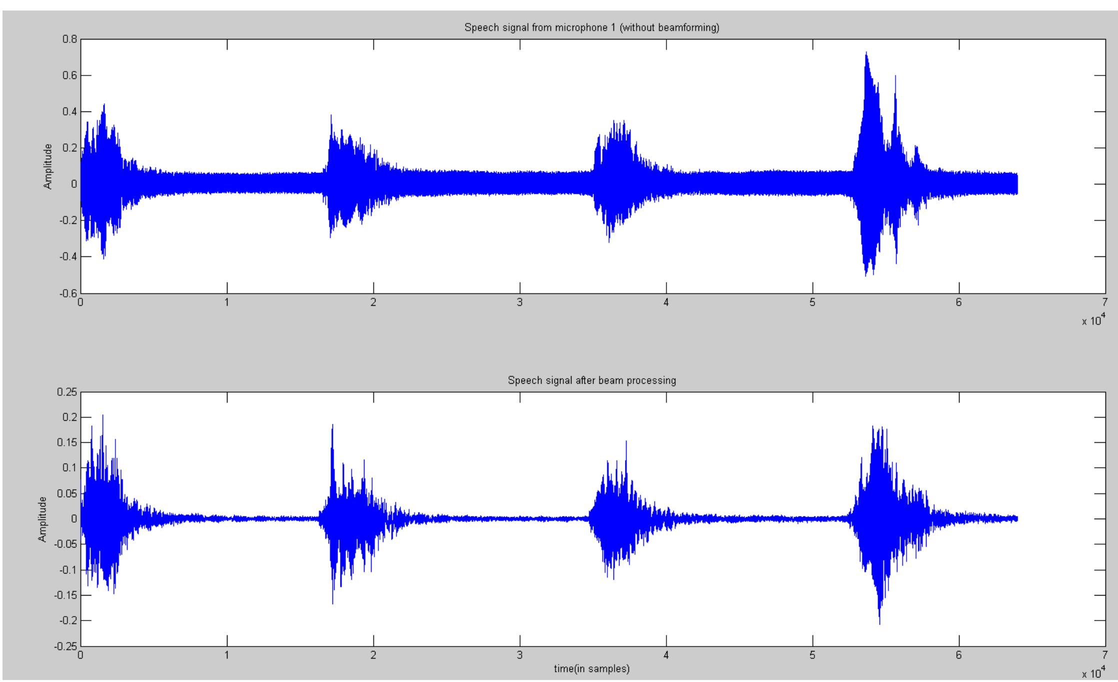

Fig.5: speech signal results with surrounding interference in lab

We can observe the attenuation of surrounding interference signals and the speech signal is unaltered as it was blocked completely by the blocking matrix.

2. To check the interference of the other person speech signal we made set up with 0 degree source speech signal and 45 degree for other speech signal.

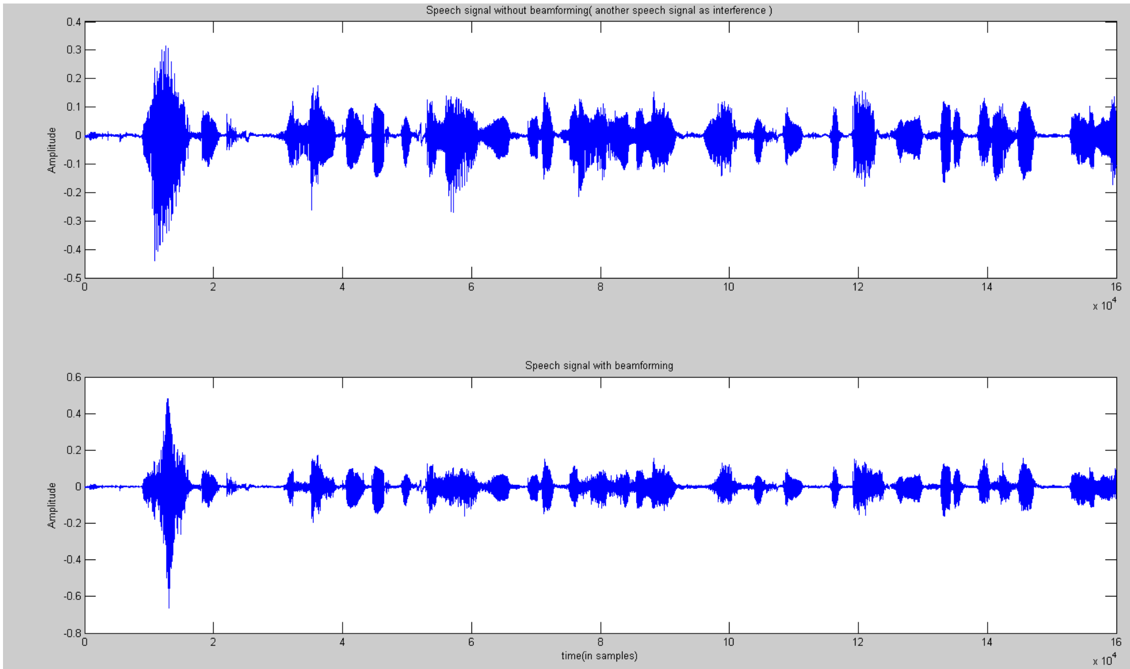

Fig.6: speech signal results with another speech signal as interference

We observed that the speech signal is having a echo effect after processing was mainly because the delayed versions of the interference signals are used to estimate the actual interference signal. So echo cancellation with the adaptive beamforming is the usual system used for the hands free telephony in car.

Other problems usually observed with this method are slight errors in the estimation of the source angle leads to cancellation of the target speech signal.

Conclusion

The adaptive beamformer using generalized side lobe canceller equipped with an adaptive blocking matrix using coefficient-constrained adaptive filters and a multiple input canceller using norm-constrained adaptive filters is studied and tested with simulations. This uses spatial domain for interference cancellation and provides good speech enhancement under different interference signals including the speech from other person. The linear constrained algorithm is implemented using simple unconstrained algorithm by using the GSC[7]. The problems with this method include target signal cancellation due to slight errors in the target angle, echo effect on the final processed signal.

References

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