A STUDY OF ADAPTIVE BEAMFORMING TECHNIQUES USING SMART ANTENNA FOR MOBILE COMMUNICATION

A THESIS SUBMITTED IN PARTIAL FULFILLMENT
OF THE REQUIREMENTS FOR THE DEGREE OF

Master of Technology

in

Electrical Engineering

By

Shankar Ram

Department of Electrical Engineering
National Institute of Technology
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Under the Guidance of

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CERTIFICATE

This is to certify that the thesis entitled, "**A Study of Adaptive Beamforming Techniques Using Smart Antenna for Mobile Communication**" submitted by Sri **Shankar Ram** in partial fulfillment of the requirements for the award of MASTER of Technology Degree in Electrical Engineering with specialization in "**Electronics System and Communication**" at the National Institute of Technology, Rourkela (Deemed University) is an authentic work carried out by him under my supervision and guidance.

To the best of my knowledge, the matter embodied in the thesis has not been submitted to any other University/ Institute for the award of any degree or diploma.

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Acknowledgement

I would like to express my deep sense of respect and gratitude toward my supervisor Dr. Susmita Das, who not only guided the academic project work but also stood as a teacher and philosopher in realizing the imagination in pragmatic way. I want to thank her for introducing me in the field of the Smart Antenna for wireless communication. Her presence and optimism have provided an invaluable influence on my career and outlook for the future. I consider it as my good fortune to have got an opportunity to work with such a wonderful person.

I express my gratitude to Dr P.K. Nanda, Professor and Head, Department of Electrical Engineering, faculty member and staff of Department of Electrical Engineering for extending all possible help in carrying out the dissertation work directly or indirectly. They have been great source of inspiration to me and I thank them from bottom of my heart. I would like to acknowledge my institute, National Institute of Technology, Rourkela, for providing good facilities to complete my thesis work.

I would also like to take this opportunity to acknowledge my friends for their support and encouragement. Without them, it would have been very difficult for me to complete my thesis work.

I am especially indebted to my parents for their love, sacrifice and support. They are my teachers after I came to this world and have set great example for me about how to live, study and work.

At last, I would like to give thanks to God since He has given me wisdom, health and all the necessities that I need for all these years.

Shankar Ram

Contents

Abstract

Mobile radio network with cellular structure demand high spectral efficiency for minimizing number of connections in a given bandwidth. One of the promising technologies is the use of "Smart Antenna". A smart antenna is actually combination of an array of individual antenna elements and dedicated signal processing algorithm. Such system can distinguish signal combinations arriving from different directions and subsequently increase the received power from the desired user. Wireless systems that enable higher data rates and higher capacities have become the need of hour. Smart antenna technology offer significantly improved solution to reduce interference level and improve system capacity. With this technology, each user's signal is transmitted and received by the base station only in the direction of that particular user. Smart antenna technology attempts to address this problem via advanced signal processing technology called beam-forming. The advent of powerful low-cost digital signal processors (DSPs), general-purpose processors (and ASICs), as well as innovative software-based signal-processing techniques (algorithms) have made intelligent antennas practical for cellular communications systems and makes it a promising new technology. Through adaptive beam-forming, a base station can form narrower beam toward user and nulls toward interfering users.

In this thesis, both the block adaptive and sample-by-sample methods are used to update weights of the smart antenna. Block adaptive beam-former employs a block of data to estimate the optimum weight vector and is known as sample matrix inversion (SMI) algorithm. The sample-by-sample method updates the weight vector with each sample. Various sample-by-sample methods, attempted in the present study are least mean square (LMS) algorithm, constant modulus algorithm (CMA), least square constant modulus algorithm(LS-CMA) and recursive least square (RLS) algorithm. In the presence of two interfering signals and noise, both amplitude and phase comparison between desired signal and estimated output, beam patterns of the smart antennas and learning characteristics of the above mentioned algorithms are compared and analyzed. The recursive least square algorithm has the faster convergence rate; however this improvement is achieved at the expense of increase in computational complexity.

Smart antennas technology suggested in this present work offers a significantly improved solution to reduce interference levels and improve the system capacity. With this novel technology, each user's signal is transmitted and received by the base station only in the direction of that particular user. This drastically reduces the overall interference in the system. Further through adaptive beam forming, the base station can form narrower beams towards the

desired user and nulls towards interfering users, considerably improving the signal-to-interference-plus-noise ratio. It provides better range or coverage by focusing the energy sent out into the cell, multi-path rejection by minimizing fading and other undesirable effects of multi-path propagation.

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CHAPTER 1
INTRODUCTION

In the past decade, there has been a tremendous growth in the area of telecommunications. As cellular phones and the high speed Internet become more and more popular, the demand for faster and more efficient telecommunication systems has been skyrocketing. Both the increase in the number of users and the increase in high data rate transfers from the Internet have produced a huge increase in traffic. In order to handle this heavy traffic, the whole telecommunications infrastructure has been transformed in the past several years. Thousands and thousands of optical fibers were laid underground to allow high speed, wide bandwidth signal transfer. This has solved most of the problems for land-based systems. However, more and more high-speed services are now carried out in a mobile environment where data transfer is done through wireless channels. Compared to the capacity of a fiber, the capacity of a wireless link is the weakest part in the whole infrastructure. Many people have realized this and have tried different methods to increase the bandwidth of the wireless channels.

Before going any further, let us explain how wireless communications systems work [1]. In most wireless communications systems, there are two major components - base stations and the mobiles. The base station is located at the center of a coverage area called a 'cell' and the mobiles can be anywhere within the cell. Communication then takes place between the base station and the mobile through the wireless channel. A certain amount of spectrum is assigned to a cell for signal transfer. This spectrum provides the media for the signals to be transmitted. With a wider spectrum, more users can be served within a cell. To serve a large area, one can use a high power base station to cover the whole area. but only a fixed amount of users can be served in this way. So instead of having a single large cell, multiple cells with smaller size are usually used to cover a large area. To avoid severe interference between cells, base stations that are adjacent to each other are allocated different spectrum or channel groups. Also the output power of the base station is maintained at a level so it is just enough to cover the whole area up to the cell boundary. With this method, cells that are separated by a certain amount of distance can use the same spectrum. This is true as long as the interference caused by surrounding co-channel cells is within tolerable limits. Figure1.1. Shows one possible frequency reuse topology with a cluster size of seven. In this frequency reuse scheme, each cell in the cluster has one-seventh of the total spectrum, and the same spectrum is used in cells with the same label. With this cellular reuse method, the same spectrum can now be reused over and over again and can be used to cover an infinitely large area.

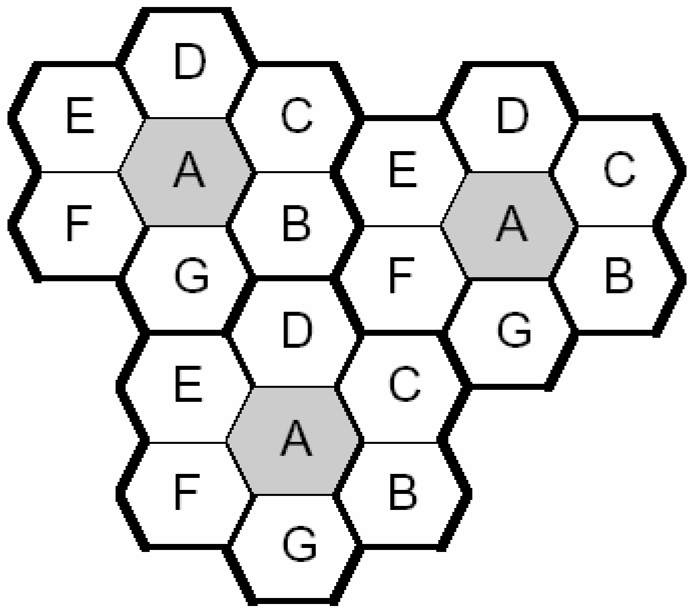

Fig.1.1. Spectrum allocation in multiple cells with frequency reuse

To allow simultaneous communication with multiple users within the same cell, the system has to have methods to isolate the signals from each user. A typical method is to separate users by frequency. In this kind of system, signals from each user are transmitted at different frequencies and are separated by proper filtering.

By now, one should have a brief idea on how wireless communications systems function. So let us explore some of the possibilities in improving the system performance. The easiest way is to use more spectrum in each cell so that more users can be handled by the same base station, provided that the user's bandwidth remains unchanged, or more bandwidth can be assigned to each user so that faster data transfer rates can be achieved through the wireless channel. Although this approach does not require any changes in the system architecture, it is an impractical and expensive approach. This is because there is limited spectrum allocated for wireless applications and it costs a lot of money for the service providers to rent it. More spectrum is available at higher frequencies, but the cost of building RF components that work at these frequency ranges becomes an issue. Another approach to enhance the capacity and the performance of the system is to minimize the size of a cell so that each serves a smaller area with the same amount of users, increasing overall density. This is a good method and is used by most service providers to increase the system capacity. However, there is still a limitation in how close co-channel cells can be put together. Besides, more base stations are required to cover the same area, driving up the cost. The last approach is quite different from the previous two. Instead of finding more spectrums to serve the same number of users, different transmission techniques are used so that the system can serve more users with the same amount of spectrum. Unlike the prior two approaches that do not require changes in the system design, this method needs hardware customized to the transmission method. Although the base station is more expensive, the additional cost is likely less than that of buying more spectrums. This system makes better use of the available spectrum and has a higher spectral efficiency.

Many transmission methods have been developed in the past for wireless communications. They vary from the very simple that support one user per channel to those that handle multiple users per channel. Some access schemes that are commonly used in wireless communications are listed below.

The simplest access scheme is called Frequency Division Multiple Access (FDMA). It is widely used in many wireless applications like TV broadcast, radio broadcast, cellular system, etc. In FDMA, signals from different users are transmitted with different frequencies just to avoid interference. Each signal is isolated from others by passing the received signals through a band pass filter. This method has been available for many decades and is used mainly in analog systems. With the invention of digital technology, new methodologies became available in designing communications systems and access schemes. The two most well known access schemes that make use of the digital technology are Time Division Multiple Access (TDMA) and Code Division Multiple Access (CDMA). Unlike FDMA, which can only support one user per channel, both of these support multiple users per channel and increase the system capacity.

In TDMA systems, users are not just separated by frequency, but are separated both by time and frequency. By doing so, multiple users can now share the same frequency channel by having a different time slot. This kind of access scheme works well for voice communication. Although a user is not transmitting or receiving during the whole communication, the signal segments are carefully put together to reconstruct the original signal at the receiver with no difference to a human ear. However, the data rate will be compromised in case of data transmission with TDMA compared to other access schemes that do not require time sharing.

Unlike TDMA, which is still partly based on the FDMA concept, CDMA uses a totally different concept. In CDMA, the signal from each user is spread across the entire channel by a unique spreading code. The codes are chosen so that they are orthogonal to each other. They act as the key to access the message in the transmitted signal. To retrieve a particular message at the receiver, the corresponding code or key is used to extract the signal. To further illustrate how the multiple access takes place, a simple analogy is used below to explain the process. In CDMA, each code acts like a mask, which makes only the signal being transmitted with the same mask visible but not signals with other masks. In an ideal situation, an infinite number of users can be added to the same channel as long as the system can supply an uncorrelated code for each signal. However, the orthogonal characteristics of the code diminish when they travel through the wireless channel. This means that the masks are not perfect and do not give perfect isolation from other signals. Leaking signals from other users now cause interference and noise in the desired signal. More interference is received if more users are added to the channel, decreasing the signal-to-interference plus noise ratio (SINR) of the desired signal. Since a certain amount of SINR is required to ensure an acceptable signal quality, this interference in turn limits the system capacity in the CDMA system. But if there is a method to minimize this co-channel interference, the system capacity can obviously be increased in the CDMA system.

TDMA or CDMA systems can have 3 to 6 times the capacity of FDMA systems. However, there is still a lot of room for improving system performance. It is always a challenge to provide higher system capacity in the network to fulfill the growing demand. A lot of research is being performed to look for methods to further improve system performance and the spectral efficiency. One recent development in this area is a use of smart antenna with algorithm [2, 3], also known as Space Division Multiple Access (SDMA) technology.

SDMA [4] is a totally different concept compared to TDMA and CDMA. It uses a different technique to separate users. Instead of using time or code, users are now separated by their spatial locations. Although SDMA may be new to commercial use, this is not the case for military applications. In the past, similar systems were used to counter electronic jamming in electronic warfare. It may not be as powerful as the one that is built in this project, but the basic concept stays the same. It makes use of the fact that the jammer is usually located at a different point to the desired communication partner. By adjusting the radiation pattern of the antenna structure, the effect of the jammer can be reduced by placing a null in the direction of the jammer. This was sometimes done in the past by manually changing the orientation of a highly directional antenna. The same concept is now used in the commercial system. But instead of eliminating jammers, the system is now used to reduce interference from co-channel users. Also, no more hand-tuning is needed in the antenna since the radiation pattern can be manipulated by software.

To implement an SDMA system, the most obvious difference in the system architecture compared to regular systems is the antenna design. In a regular base station, the antenna is either omni-directional or sectorized. To establish a connection between the base station and the mobile, the transmitted signal is broadcast from the base station antenna. Its goal is to cover the entire cell or sector so that the mobile can pick up the broadcast signal as long as it is within the coverage area. However, this kind of signal transmission wastes a lot of energy due to the fact that the mobile being addressed can only occupy one spot at a time. With broadcasting, most of the power is radiated in other directions instead of traveling toward the desired user. Besides, the broadcasting signal also causes undesired interference to other users located within the cell. If there is a way to pinpoint a user within the cell so that the transmitting antenna aims toward the desired user, less power will be needed to carry out the transmission. Also a lot of unnecessary interference can be removed from the desired signal. This is the basic philosophy behind SDMA.

One may wonder, if this concept has been around for so many years and is so beneficial, why has it only recently become popular. There is no simple answer to this question, but the computation requirement in such a system plays a major role in hindering its implementation. As mentioned earlier, in order for the system to have a flexible radiation pattern, some kind of adaptation process has to be added to the system. There are different levels of intelligence available and the kind of intelligence selected directly affects the performance of the system. In the past, computational cost was large. The computation requirement needed to implement the simplest algorithm could cost quite a lot of money. Also, there was no reason for the provider to spend money on something that was not necessary at that time. Only recently has the demand for system capacity gone up tremendously so that sophisticated multiple access schemes are now economically viable.

In order to manipulate the radiation pattern of an antenna structure with software, multiple antennas are required instead of a single antenna. Unlike a single antenna, which has a fixed radiation pattern, the radiation pattern of an antenna array can be quite flexible. The flexibility varies according to the algorithm being implemented in the system. The most straight forward approach to generate a flexible radiation pattern is the switched lobe (SL) or the switched beam technique where the antenna array contains a number of highly directional antennas. Each of the antennas points in a slightly different direction. The system then analyzes the received signal from each of the antennas and selects the one that has the best signal. A more intelligent approach would be, instead of switching antennas, determine the direction of arrival (DoA) of the signal. Once the DoA is obtained, the system uses the antenna array to form a highly directional beam pointing toward the user. Both methods should provide some advantages over the conventional system; however the benefit would be minimal if the signal suffers a lot of angular spread where the signal arrives at many different directions in a multipath environment. The situation would be even worse when no line-of-sight (LOS) is present between the user and the base station.

To overcome the above shortcoming, a more advanced method was developed. This method, usually called the optimum beam forming technique, fully utilizes the spatial diversity present in the multipath channel so that a stronger received signal can be generated. With optimum beam forming, signals received from multiple antennas are adjusted separately in both amplitude and phase before being combined. By doing so, the system behaves as if it has multiple adjustable radiation patterns. Each of the patterns is tuned to receive signals from a single user. An adaptive algorithm is used at the base station so that the system has the ability to determine the optimal radiation pattern for each user. As part of the training procedure, each of the users transmits a short training sequence to the base station. The algorithm then makes use of this information from a user by comparing each received signal to the original sequence to find out the correct radiation pattern for that user. With this method, all received signals from each antenna element are used and are optimally combined to enhance the desired signal and to cancel unwanted interference. During the training process, a lot of number crunching is needed at the base station. So it was not popular in the past due to the expensive cost of computation power. However, intensive signal processing is no longer an issue with the availability of low cost, extremely fast processors. Keep in mind that what actually happens in optimal beam forming is more complicated than what is shown in the diagram. It is more complicated when interference from other mobile occurs.

As mentioned earlier, SDMA provides a lot of benefits. For example, signals are now radiated more directly toward the selected user. This increases the power efficiency and extends the coverage area. Multipath fading is no longer an issue with a SDMA system. With multiple antennas, each of them located at a different spot, the possibility of having a fade at all of the antennas simultaneously is very small and can be ignored. Interference from other users is now minimized. This improves the SINR of the received signal and therefore gives better signal quality. It also provides another way of having multiple users on the same frequency channel since signals originating from users at different locations are reduced after the signal combining. This further increases the system capacity and also the spectral efficiency.

The development of a truly personal communication space will rely on the design of next-generation wireless system which require adoption of smart antenna technique in order to provide the expected beneficial impact on efficient use of the spectrum, minimization of the cost, robust and transparent operation across multi-technology wireless network. Today, when spectrally efficient solutions are increasingly a business imperative, these systems are providing greater coverage area for each cell site, higher rejection of interference, data rate and substantial capacity improvements as shown in fig1.2.

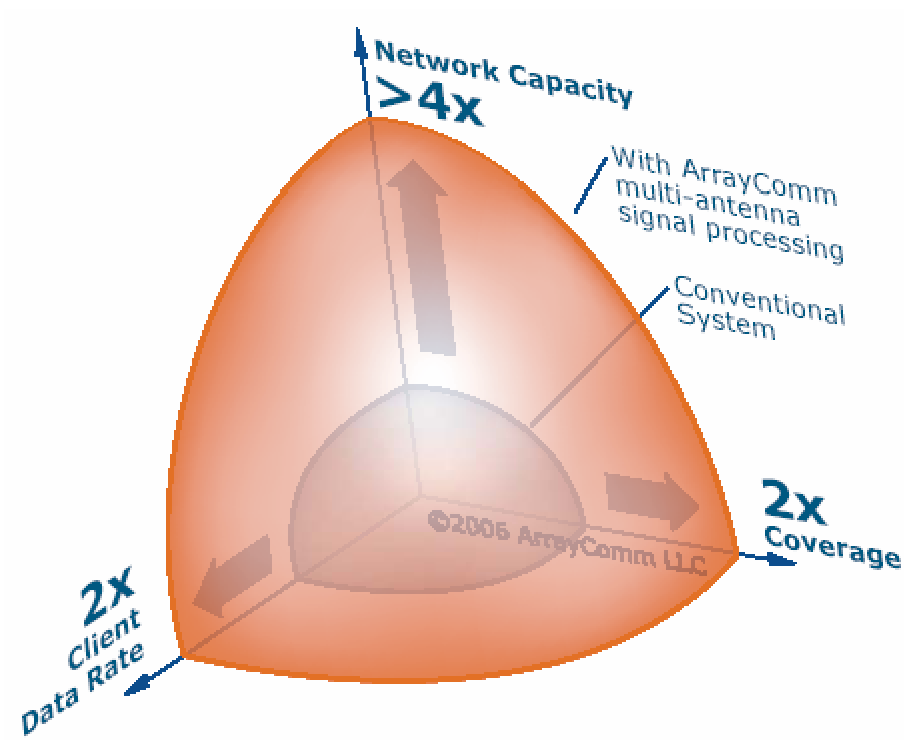

Fig.1.2. Impact of smart antenna on wireless communication

CHAPTER 2

ANTENNAS AND ANTENNA SYSTEMS

2.1 A Useful Analogy for Adaptive Smart Antenna

For an intuitive grasp of how an adaptive antenna system works, close your eyes and converse with someone as they move about the room. You will notice that you can determine their location without seeing them because of the following:

- You hear the speaker's signals through your two ears, your acoustic sensors.
- The voice arrives at each ear at a different time.
- Your brain, a specialized signal processor, does a large number of calculations to correlate information and compute the location of the speaker.

Your brain also adds the strength of the signals from each ear together, so you perceive sound in one chosen direction as being twice as loud as everything else.

Adaptive antenna systems [5] do the same thing, using antennas instead of ears. As a result, 8, 10, or 12 ears can be employed to help fine-tune and turn up signal information. Also, because antennas both listen and talk, an adaptive antenna system can send signals back in the same direction from which they came. This means that the antenna system cannot only hear 8 or 10 or 12 times louder but talk back more loudly and directly as well.

Going a step further, if additional speakers joined in, your internal signal processor could also tune out unwanted noise (interference) and alternately focus on one conversation at a time. Thus, advanced adaptive array systems have a similar ability to differentiate between desired and undesired signals.

2.2 Antennas

Radio antennas couple electromagnetic energy from one medium (space) to another (e.g., wire, coaxial cable, or waveguide). Physical designs can vary greatly.

2.2.1 Omni Directional Antennas

Since the early days of wireless communications, there has been the simple dipole antenna, which radiates and receives equally well in all directions. To find its users, this single-element design broadcasts omni directionally in a pattern resembling ripples radiating outward in a pool of water. While adequate for simple RF environments where no specific knowledge of the users' whereabouts is available, this unfocused approach scatters signals, reaching desired users with only a small percentage of the overall energy sent out into the environment.

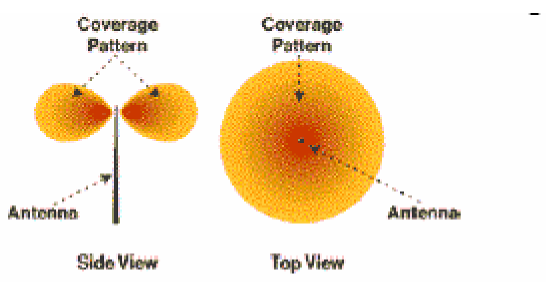

Fig.2.1. Omni directional Antenna and Coverage Patterns

Given this limitation, omni directional strategies attempt to overcome environmental challenges by simply boosting the power level of the signals broadcast. In a setting of numerous users (and interferers), this makes a bad situation worse in that the signals that miss the intended user become interference for those in the same or adjoining cells.

In uplink applications (user to base station), omni directional antennas offer no preferential gain for the signals of served users. In other words, users have to shout over competing signal energy. Also, this single-element approach cannot selectively reject signals interfering with those of served users and has no spatial multi-path mitigation or equalization capabilities.

Omni directional strategies directly and adversely impact spectral efficiency, limiting frequency reuse. These limitations force system designers and network planners to devise increasingly sophisticated and costly remedies. In recent years, the limitations of broadcast antenna technology on the quality, capacity, and coverage of wireless systems have prompted an evolution in the fundamental design and role of the antenna in a wireless system.

2.2.2 Directional Antennas

A single antenna can also be constructed to have certain fixed preferential transmission and reception directions [5]. As an alternative to the brute force method of adding new transmitter sites, many conventional antenna towers today split, or sectaries cells. A 360° area is often split into three 120° subdivisions, each of which is covered by a slightly less broadcast method of transmission.

All else being equal, sector antennas provide increased gain over a restricted range of azimuths as compared to an omni directional antenna. This is commonly referred to as antenna element gain and should not be confused with the processing gains associated with smart antenna systems. While sectaries antennas multiply the use of channels, they do not overcome the major disadvantages of standard omni directional antenna broadcast such as co channel Interference.

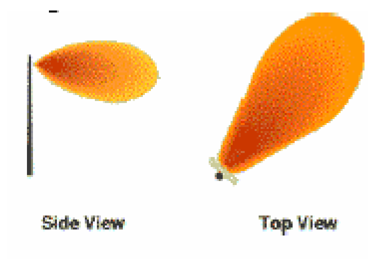

Fig.2.2. Directional Antenna and Coverage Pattern

2.3 Antenna Systems

How can an antenna be made more intelligent? First, its physical design can be modified by adding more elements. Second, the antenna can become an antenna system that can be designed to shift signals before transmission at each of the successive elements so that the antenna has a composite effect. This basic hardware and software concept is known as the phased array antenna.

The following summarizes antenna developments in order of increasing benefits and intelligence.

2.3.1 Sectorized Systems

Sectorized antenna systems [5] take a traditional cellular area and subdivide it into sectors that are covered using directional antennas looking out from the same base station location. Operationally, each sector is treated as a different cell, the range of which is greater than in the omni directional case. Sector antennas increase the possible reuse of a frequency channel in such cellular systems by reducing potential interference across the original cell, and they are widely used for this purpose. As many as six sectors per cell have been used in practical service. When combining more than one of these directional antennas, the base station can cover all directions.

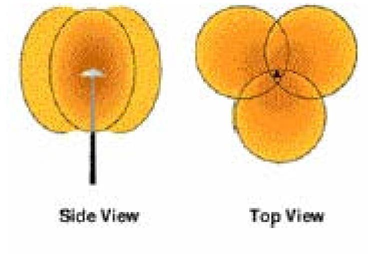

Fig.2.3. Sectorized Antenna and Coverage Patterns

2.3.2 Diversity Systems

In the next step toward smart antennas, the diversity system [6] incorporates two antenna elements at the base station, the slight physical separation (space diversity) of which has been used historically to improve reception by counteracting the negative effects of multipath .

Diversity offers an improvement in the effective strength of the received signal by using one of the following two methods:

- **Switched diversity**. Assuming that at least one antenna will be in a favorable location at a given moment, this system continually switches between antennas (connects each of the receiving channels to the best serving antenna) so as always to use the element with the largest output. While reducing the negative effects of signal fading, they do not increase gain since only one antenna is used at a time.
- **Diversity combining**. This approach [7] corrects the phase error in two multipath signals and effectively combines the power of both signals to produce gain. Other diversity systems, such as maximal ratio combining systems, combine the outputs of all the antennas to maximize the ratio of combined received signal energy to noise.

Because macro cell-type base stations historically put out far more power on the downlink (base station to user) than mobile terminals can generate on the reverse path, most diversity antenna systems have evolved only to perform in uplink (user to base station).

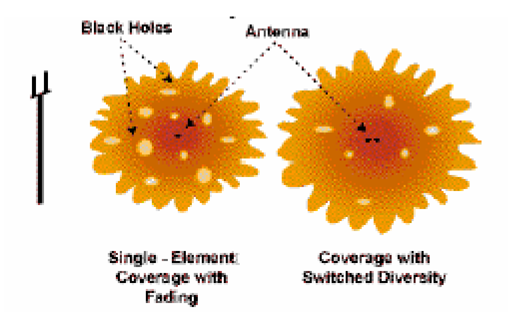

Fig. 2.4. Switched Diversity Coverage with Fading and Switched Diversity

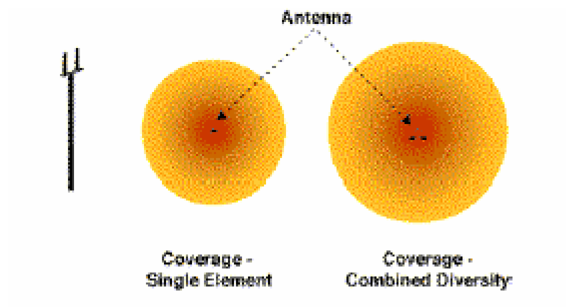

Fig2.5. Combined Diversity Effective Coverage Pattern with Single Element and Combined Diversity

Diversity antennas merely switch operation from one working element to another. Although this approach mitigates severe multipath fading, its use of one element at a time offers no uplink gain improvement over any other single element approach. In high-interference environments, the simple strategy of locking onto the strongest signal or extracting maximum signal power from the antennas is clearly inappropriate and can result in crystal-clear reception of an interferer rather than the desired signal.

The need to transmit to numerous users more efficiently without compounding the interference problem led to the next step of the evolution antenna systems that intelligently integrate the simultaneous operation of diversity antenna elements.

2.3.3 Smart

The concept of using multiple antennas and innovative signal processing to serve cells more intelligently has existed for many years. In fact, varying degrees of relatively costly smart antenna [2, 3, 5] systems have already been applied in defense systems. Until recent years, cost barriers have prevented their use in commercial systems. The advent of powerful low-cost digital signal processors(DSPs), general-purpose processors (and ASICs), as well as innovative software-based signal-processing techniques (algorithms) have made intelligent antennas practical for cellular communications systems.

Today, when spectrally efficient solutions are increasingly a business imperative, these systems are providing greater coverage area for each cell site, higher rejection of interference, and substantial capacity improvements.

CHAPTER 3

BRIEF OVERVIEW OF SMART ANTENNA SYSTEM

In truth, antennas are not smart—antenna systems are smart. Generally collocated with a base station, a smart antenna system combines an antenna array with a digital signal-processing capability to transmit and receive in an adaptive, spatially sensitive manner. In other words, such a system can automatically change the directionality of its radiation patterns in response to its signal environment. This can dramatically increase the performance characteristics (such as capacity) of a wireless system.

3.1 How Many Types of Smart Antenna Systems Are There?

Terms commonly heard today that embrace various aspects of a smart antenna system technology include intelligent antennas, phased array, SDMA, spatial processing, digital beam forming, adaptive antenna systems, and others. Smart antenna systems [8] are customarily categorized, however, as either switched beam or adaptive array systems. The following are distinctions between the two major categories of smart antennas regarding the choices in transmit strategy:

- **Switched beam**. A finite number of fixed, predefined patterns or combining strategies (sectors)
- **Adaptive array**. An infinite number of patterns (scenario-based) that are adjusted in real time.

3.1.1 What Are Switched Beam Antennas?

Switched beam antenna systems form multiple fixed beams with heightened sensitivity in particular directions. These antenna systems detect signal strength, choose from one of several predetermined, fixed beams, and switch from one beam to another as the mobile moves throughout the sector. Instead of shaping the directional antenna pattern with the metallic properties and physical design of a single element (like a sectorized antenna), switched beam systems combine the outputs of multiple antennas in such a way as to form finely sectorized (directional) beams with more spatial selectivity than can be achieved with conventional, single-element approaches.

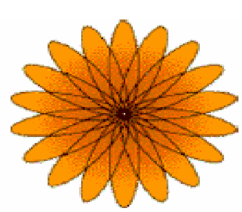

Fig.3.1. Switched Beam System Coverage Patterns (Sectors)

3.1.2 What Are Adaptive Array Antennas?

Adaptive antenna technology represents the most advanced smart antenna approach to date. Using a variety of new signal-processing algorithms, the adaptive system takes advantage of its ability to effectively locate and track various types of signals to dynamically minimize interference and maximize intended signal reception.

Both systems attempt to increase gain according to the location of the user; however, only the adaptive system provides optimal gain while simultaneously identifying, tracking, and minimizing interfering signals.

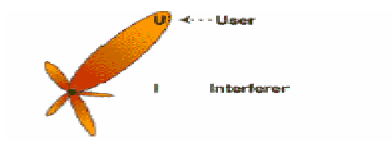

Fig3.2. Adaptive Array Coverage.

3.2 What Do They Look Like?

Omni directional antennas are obviously distinguished from their intelligent counterparts by the number of antennas (or antenna elements) employed. Switched beam and adaptive array systems, however, share many hardware characteristics and are distinguished primarily by their adaptive intelligence.

To process information that is directionally sensitive requires an array of antenna elements (typically 4 to 12), the inputs from which are combined to control signal transmission adaptively. Antenna elements can be arranged in linear, circular, or planar configurations and are most often installed at the base station, although they may also be used in mobile phones or laptops.

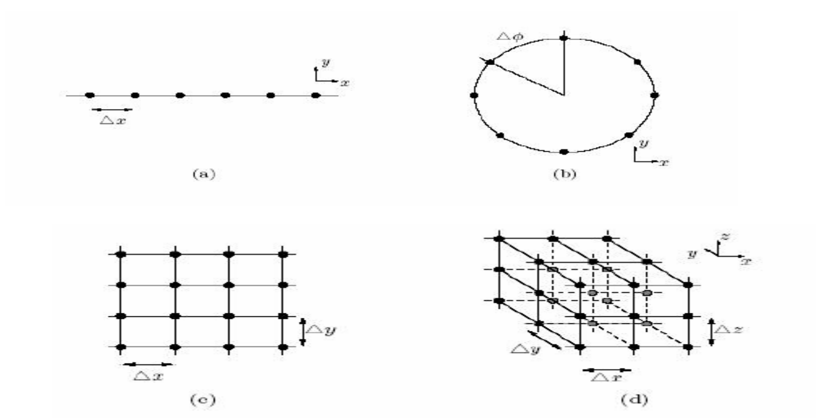

Fig.3.3.Different array geometries for smart antenna: a) Uniform linear array, b) Circular array, c) Two dimensional grid array and d) Three dimensional grid array

3.3 The Goals of the Smart Antenna System

The dual purpose of a smart antenna system is to augment the signal quality of the radio-based system through more focused transmission of radio signals while enhancing capacity through increased frequency reuse. More specifically, the features of and benefits derived from a smart antenna system include those listed in table 3.1

Feature	**Benefit**
Signal gain-Inputs from multiple antennas are combined to optimize available power required to establish given level of coverage.	**Better range/coverage**-Focusing the energy sent out into the cell increases base station range and coverage. Lower power requirements also enable a greater battery life and smaller/lighter handset size.
Interference rejection- Antenna pattern can be generated toward co channel interference sources, improving the signal- to interference ratio of the received signals.	**Increased capacity**- Precise control of signal nulls quality and mitigation of interference combine to frequency reuse reduce distance (or cluster size), improving capacity. Certain adaptive technologies (such as space division multiple access) support the reuse of frequencies within the same cell.
Spatial diversity-Composite information from the array is used to minimize fading and other undesirable effects of multipath propagation.	**multipath rejection**-can reduce the effective delay spread of the channel, allowing higher bit rates to be supported without the use of an equalizer
Power efficiency-Combines the inputs to multiple elements to optimize available processing gain in the downlink(toward users)	**reduced expense**-Lower amplifier costs, power consumption, and higher reliability will result

3.4 A few drawbacks of Smart Antenna

Smart-antenna transceivers are much more complex than traditional base-station transceivers. The antenna array needs separate transceiver chains for each antenna element in the array, and accurate real-time calibration for each of them. Moreover, the antenna beam forming is computationally intensive, which means that smart-antenna base stations must be equipped with very powerful digital signal processors. This tends to increase the system costs in the short term; however, since the benefits outweigh the costs, it will be cheaper in the long run.

For a smart antenna to have a reasonable gain, an array of antenna elements is necessary. Consequently, this means that a linear array consisting of 10 elements with an inter-element spacing of $\lambda/2$, operating at 2 GHz, would be approximately 70 cm wide. This might pose problems, due to the growing public demand for less-visible base stations.

CHAPTER 4

MULTIPATH AND CO CHANNEL INTERFERENCE

4.1 A Useful Analogy for Signal Propagation

Envision a perfectly still pool of water into which a stone is dropped. The waves that radiate outward from that point are uniform and diminish in strength evenly. This pure omni directional broadcasting equates to one caller's signal—originating at the terminal and going uplink. It is interpreted as one signal everywhere it travels.

Picture now a base station at some distance from the wave origin. If the pattern remains undisturbed, it is not a challenge for a base station to interpret the waves. But as the signal's waves begin to bounce off the edges of the pool, they come back (perhaps in a combination of directions) to intersect with the original wave pattern. As they combine, they weaken each other's strength. These are multipath interference problems.

Now, picture a few more stones being dropped in different areas of the pool, equivalent to other calls starting. How could a base station at any particular point in the pool distinguish which stone's signals were being picked up and from which direction? This multiple-source problem is called co channel interference.

These are two-dimensional analogies; to fully comprehend the distinction between callers and/or signal in the earth's atmosphere, a base station must possess the intelligence to place the information it analyzes in a true spatial context.

4.2 Multipath

Multipath is a condition where the transmitted radio signal is reflected by physical features/structures, creating multiple signal paths between the base station and the user terminal.

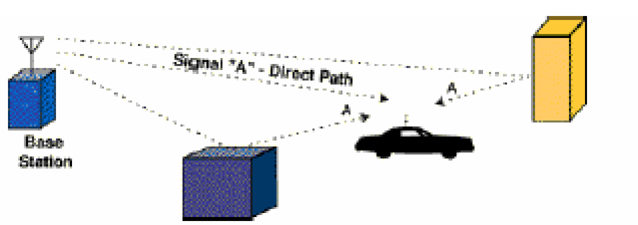

Fig.4.1. The Effect of Multipath on a Mobile User

4.3 Problems Associated with Multipath

One problem resulting from having unwanted reflected signals is that the phases of the waves arriving at the receiving station often do not match. The phase of a radio wave is simply an arc of a radio wave, measured in degrees, at a specific point in time.Fig4.2. illustrates two out-of-phase signals as seen by the receiver.

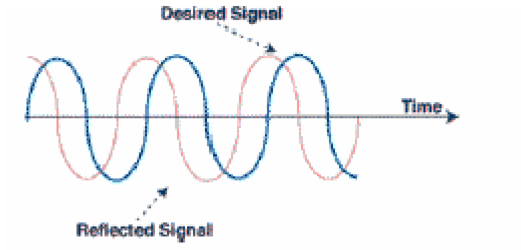

Figure 4.2. Two Out-of-Phase Multipath Signals

Conditions caused by multipath that are of primary concern are as follows:

 fading —When the waves of multipath signals are out of phase, reduction in signal strength can occur. One such type of reduction is called a fade; the phenomenon is known as "Rayleigh fading" or "fast fading."

A fade is a constantly changing, three-dimensional phenomenon. Fade zones tend to be small, multiple areas of space within a multipath environment that cause periodic attenuation of a received signal for users passing through them. In other words, the received signal strength will fluctuate downward, causing a momentary, but periodic, degradation in quality.

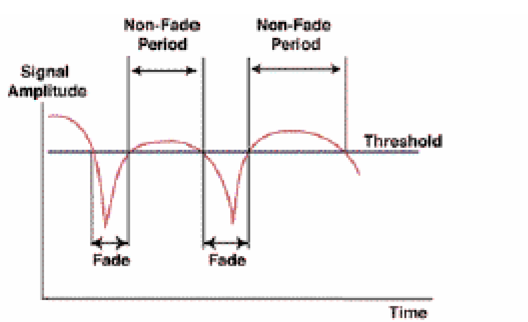

Fig. 4.3. A Representation of the Rayleigh Fade Effect on a User Signal

 phase cancellation —When waves of two multipath signals are rotated to exactly 180° out of phase, the signals will cancel each other. While this sounds severe, it is rarely sustained on any given call (and most air interface standards are quite resilient to phase cancellation). In other words, a call can be maintained for a certain period of time while there is no signal, although with very poor quality. The effect is of more concern when the control channel signal is canceled out, resulting in a black hole, a service area in which call set-ups will occasionally fail.

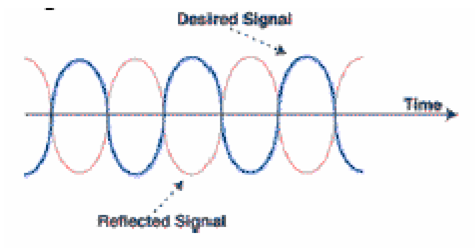

Fig.4.4. Illustration of Phase Cancellation

 delay spread— The effect of multipath on signal quality for a digital air interface (e.g., TDMA) can be slightly different. Here, the main concern is that multiple reflections of the same signal may arrive at the receiver at different times. This can result in inter symbol interference (or bits crashing into one another) that the receiver cannot sort out. When this occurs, the bit error rate rises and eventually causes noticeable degradation in signal quality.

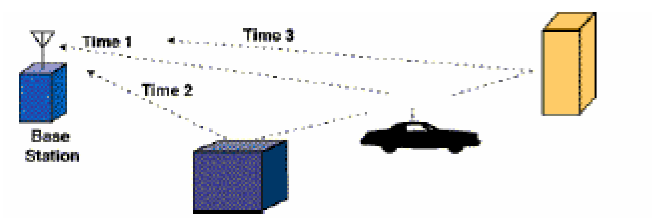

Fig.4.5. Multipath: The Cause of Delay Spread

While switched diversity and combining systems do improve the effective strength of the signal received, their use in the conventional macro cell propagation environment has been typically reverse-path limited due to a power imbalance between base station and mobile unit. This is because macro cell-type base stations have historically put out far more power than mobile terminals were able to generate on the reverse path.

 co channel interference —One of the primary forms of man-made signal degradation associated with digital radio, co channel interference occurs when the same carrier frequency reaches the same receiver from two separate transmitters.

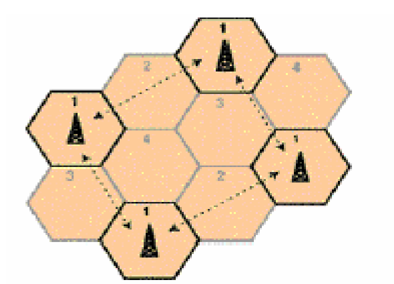

Fig.4.6. Illustration of Co channel Interference in a Typical Cellular Grid

As we have seen, both broadcast antennas as well as more focused antenna systems scatter signals across relatively wide areas. The signals that miss an intended user can become interference for users on the same frequency in the same or adjoining cells.

While sectorized antennas multiply the use of channels, they do not overcome the major disadvantage of standard antenna broadcast—co channel interference. Management of co channel interference is the number-one limiting factor in maximizing the capacity of a wireless system. To combat the effects of co channel interference, smart antenna systems not only focus directionally on intended users, but in many cases direct nulls or intentional noninterference toward known, undesired users

CHAPTER 5

THE ARCHITECTURE OF SMART ANTENNA SYSTEM

5.1 How Do Smart Antenna Systems Work?

Traditional switched beam and adaptive array systems enable a base station to customize the beams they generate for each remote user effectively by means of internal feedback control. Generally speaking, each approach forms a main lobe toward individual users and attempts to reject interference or noise from outside of the main lobe.

5.1.1 Listening to the Cell (Uplink Processing)

It is assumed here that a smart antenna is only employed at the base station and not at the handset or subscriber unit. Such remote radio terminals transmit using omni directional antennas, leaving it to the base station to separate the desired signals from interference selectively.

Typically, the received signal from the spatially distributed antenna elements is multiplied by a weight, a complex adjustment of amplitude and a phase. These signals are combined to yield the array output. An adaptive algorithm controls the weights according to predefined objectives. For a switched beam system, this may be primarily maximum gain; for an adaptive array system, other factors may receive equal consideration. These dynamic calculations enable the system to change its radiation pattern for optimized signal reception.

5.1.2 Speaking to the Users (Downlink Processing)

The task of transmitting in a spatially selective manner is the major basis for differentiating between switched beam and adaptive array systems. As described below, switched beam systems communicate with users by changing between preset directional patterns, largely on the basis of signal strength. In comparison, adaptive arrays attempt to understand the RF environment more comprehensively and transmit more selectively.

The type of downlink processing used depends on whether the communication system uses time division duplex (TDD), which transmits and receives on the same frequency or frequency division duplex (FDD), which uses separate frequencies for transmit and receiving (e.g., GSM). In most FDD systems, the uplink and downlink fading and other propagation characteristics may be considered independent, whereas in TDD systems the uplink and downlink channels can be considered reciprocal. Hence, in TDD systems uplink channel information may be used to achieve spatially selective transmission. In FDD systems, the uplink channel information cannot be used directly and other types of downlink processing must be considered.

5.2 Switched Beam Systems

In terms of radiation patterns, switched beam [9] is an extension of the current microcellular or cellular sectorization method of splitting a typical cell. The switched beam approach further subdivides macro sectors into several micro sectors as a means of improving range and capacity. Each micro sector contains a predetermined fixed beam pattern with the greatest sensitivity located in the center of the beam and less sensitivity elsewhere. The design of such systems involves high-gain, narrow azimuthally beam width antenna elements.

The switched beam system selects one of several predetermined fixed-beam patterns (based on weighted combinations of antenna outputs) with the greatest output power in the remote user's channel. These choices are driven by RF or base band DSP hardware and software. The system switches its beam in different directions throughout space by changing the phase differences of the signals used to feed the antenna elements or received from them. When the mobile user enters a particular macro sector, the switched beam system selects the micro sector containing the strongest signal. Throughout the call, the system monitors signal strength and switches to other fixed micro sectors as required.

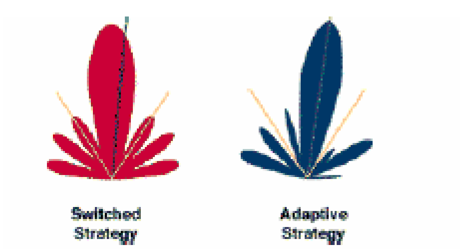

Fig.5.1. Beam forming Lobes and Nulls that Switched Beam (Red) and Adaptive Array (Blue) Systems might choose for Identical User Signals (Green Line) and Co channel Interferers (Yellow Lines)

Smart antenna systems communicate directionally by forming specific antenna beam patterns. When a smart antenna directs its main lobe with enhanced gain in the direction of the user, it naturally forms side lobes and nulls or areas of medium and minimal gain respectively in directions away from the main lobe. Different switched beam and adaptive smart antenna systems control the lobes and the nulls with varying degrees of accuracy and flexibility.

5.3 Adaptive Antenna Approach

The adaptive antenna systems approach communication between a user and base station in a different way, in effect adding a dimension of space. By adjusting to an RF environment as it changes (or the spatial origin of signals), adaptive antenna technology can dynamically alter the signal patterns to near infinity to optimize the performance of the wireless system.

Adaptive arrays utilize sophisticated signal-processing algorithms to continuously distinguish between desired signals, multipath, and interfering signals as well as calculate their directions of arrival. This approach continuously updates its transmit strategy based on changes in both the desired and interfering signal locations. The ability to track users smoothly with main lobes and interferers with nulls ensures that the link budget is constantly maximized because there are neither micro sectors nor predefined patterns.

Figure 5.2 illustrates the relative coverage area for conventional sectorized, switched beam, and adaptive antenna systems [8]. Both types of smart antenna systems provide significant gains over conventional sectored systems. The low level of interference on the left represents a new wireless system with lower penetration levels. The significant level of interference on the right represents either a wireless system with more users or one using more aggressive frequency reuse patterns. In this scenario, the interference rejection capability of the adaptive system provides significantly more coverage than either the conventional or switched beam system.

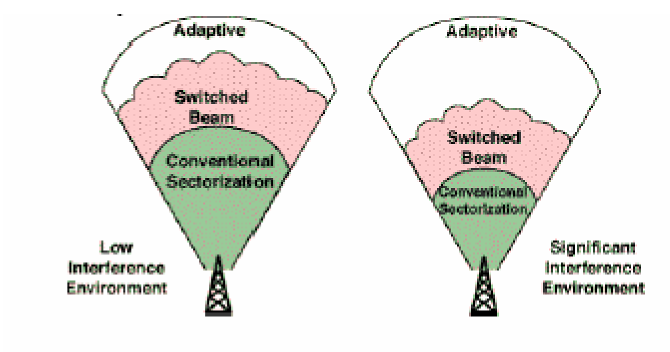

Fig.5.2. Coverage Patterns for Switched Beam and Adaptive Array Antennas

5.4 Relative Benefits/Tradeoffs of Switched Beam and Adaptive Array Systems

integration— Switched beam systems are traditionally designed to retrofit widely deployed cellular systems. It has been commonly implemented as an add-on or appliqué technology that intelligently addresses the needs of mature networks. In comparison, adaptive array systems have been deployed with a more fully integrated approach that offers less hardware redundancy than switched beam systems but requires new build-out.

 range/coverage— Switched beam systems can increase base station range from 20 to 200 percent over conventional sectored cells, depending on environmental circumstances and the hardware/software used. The added coverage can save an operator substantial infrastructure costs and means lower prices for consumers. Also, the dynamic switching from beam to beam conserves capacity because the system does not send all signals in all directions. In comparison, adaptive array systems can cover a broader, more uniform area with the same power levels as a switched beam system.

 interference suppression— Switched beam antennas suppress interference arriving from directions away from the active beam's center. Because beam patterns are fixed, however, actual interference rejection is often the gain of the selected communication beam pattern in the interferer's direction. Also, they are normally used only for reception because of the system's ambiguous perception of the location of the received signal (the consequences of transmitting in the wrong beam being obvious). Also, because their beams are predetermined, sensitivity can occasionally vary as the user moves through the sector.

Switched beam solutions work best in minimal to moderate co channel interference and have difficulty in distinguishing between a desired signal and an interferer. If the interfering signal is at approximately the center of the selected beam, and the user is away from the center of the selected beam, the interfering signal can be enhanced far more than the desired signal. In these cases, the quality is degraded for the user. Adaptive array technology currently offers more comprehensive interference rejection. Also, because it transmits an infinite, rather than finite, number of combinations, its narrower focus creates less interference to neighboring users than a switched-beam approach.

 spatial division multiple access (SDMA)—Among the most sophisticated utilizations of smart antenna technology is SDMA, which employs advanced processing techniques to, in effect, locate and track fixed or mobile terminals, adaptively steering transmission signals toward users and away from interferers. This adaptive array technology achieves superior levels of interference suppression, making possible more efficient reuse of frequencies than the standard fixed hexagonal reuse patterns. In essence, the scheme can adapt the frequency allocations to where the most users are located.

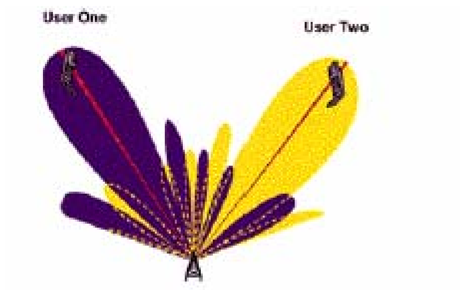

Fig.5.3. Fully Adaptive Spatial Processing, Supporting Two Users on the Same Conventional Channel Simultaneously in the Same Cell

Utilizing highly sophisticated algorithms and rapid processing hardware, spatial processing takes the reuse advantages that result from interference suppression to a new level. In essence, spatial processing dynamically creates a different sector for each user and conducts a frequency/channel allocation in an ongoing manner in real time.

Adaptive spatial processing integrates a higher level of measurement and analysis of the scattering aspects of the RF environment. Whereas traditional beam forming and beam-steering techniques assume one correct direction of transmission toward a user, spatial processing maximizes the use of multiple antennas to combine signals in space in a method that transcends a one user-one beam methodology.

CHAPTER 6

BASICS OF SMART ANTENNA APPROACH

6.1 Antenna Array (Smart Antenna)

Generally co-located with a base station, a smart antenna system combines an antenna array with a digital signal-processing capability to transmit and receive in an adaptive, spatially sensitive manner. In other words, such a system can automatically change the directionality of its radiation patterns in response to its signal environment. This can dramatically increase the performance characteristics (such as capacity) of a wireless system.

In many applications, the desired information to be extracted from an array of sensors is the content of a spatially propagating signal from a certain direction. The content may be a massage content in the signal, such as in communications applications, or merely the existence of the signal, as in the radar and sonar. To this end, we want to linearly combine the signals from all the sensors in a manner that is with a certain weighting, so as to examine signals arriving from specific angles. This operation is known as beam-forming because the weighting process emphasizes signals from a particular direction while attenuating those from other directions and can be thought of as forming a beam. In this sense, the beam-former is a spatial filter.

A standard tool for analyzing the performance of a beam-former as shown in fig.6.1 is the response for a given N-by-1 weight vector w(n) as function of θ, known as the beam response [2,3,10] . This angular response is computed for all possible angles, that is $-90^{\circ} \leq \theta \leq 90^{\circ}$

$$R(\theta) = W^{H}(n)S(\theta) \qquad (6.1)$$

Where, $S(\theta)$ is an N-by-1 steering vector? The dependence of the steering vector on θ is defined with the use of the relationship

$$S(\theta) = [1, e^{-j\theta}, e^{-2j\theta}, ... e^{-j(N-1)\theta}]^{T} \qquad (6.2)$$

Let ϕ denote the actual angle of incidence of a plane wave, measured with respect to the normal to the linear array then

$$\theta = \frac{2\pi d}{\lambda}\sin\phi \quad , \qquad -\pi/2 \leq \theta \leq \pi/2 \qquad (6.3)$$

Where d is the spacing between adjacent sensors of the array and λ is the wave length of the incident wave. First, the conventional beam-former will be discussed, and then the adaptive beam-former will be discussed.

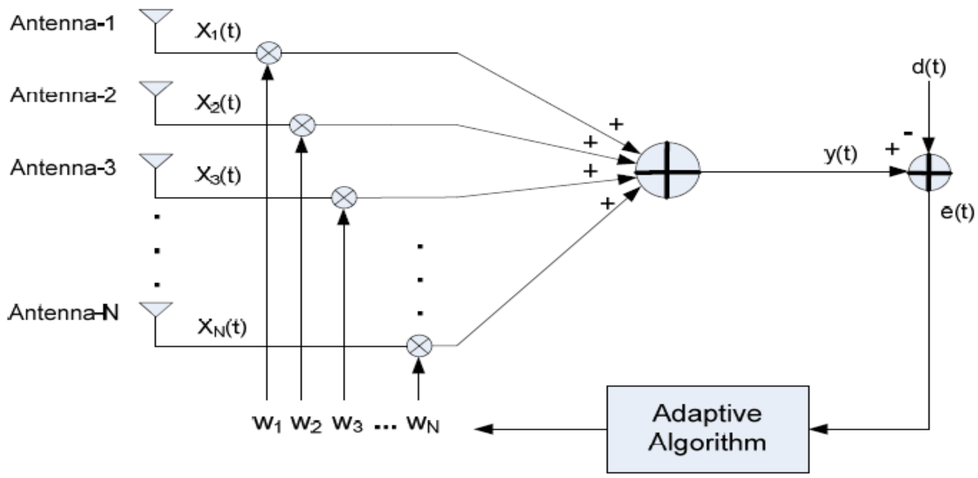

Fig.6.1. Adaptive Beam forming block diagram

6.2 Conventional Beam-forming

A conventional beam-former [2] as shown in fig6.2 is a smart antenna in which fixed weight is used to study the signal arriving from a specific direction. Since it optimize the signal arriving from specific direction while attenuating signals from other directions, thus it is called the spatial matched filter.

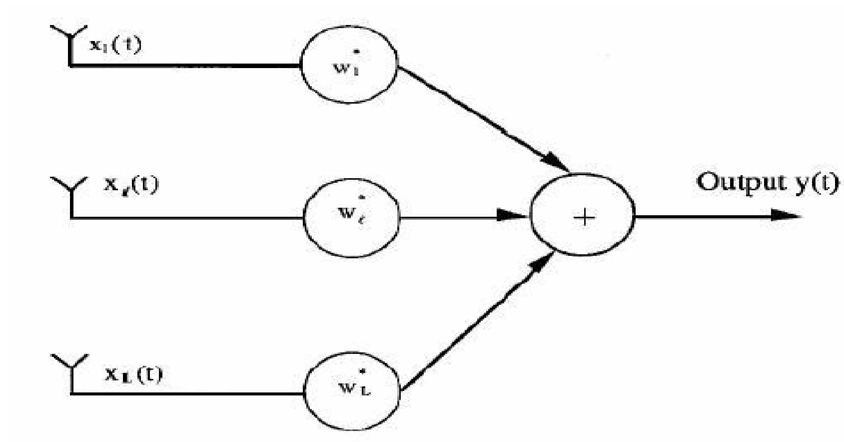

Fig6.2. Conventional beam former

A sample beam-pattern for a 16-element uniform linear array with uniform weighting ($1/\sqrt{16}$) is shown in fig.6.3, which is plotted on a logarithm scale in decibels. The large main lobe is centered at the angle 0°,the direction in which the signal is arriving.

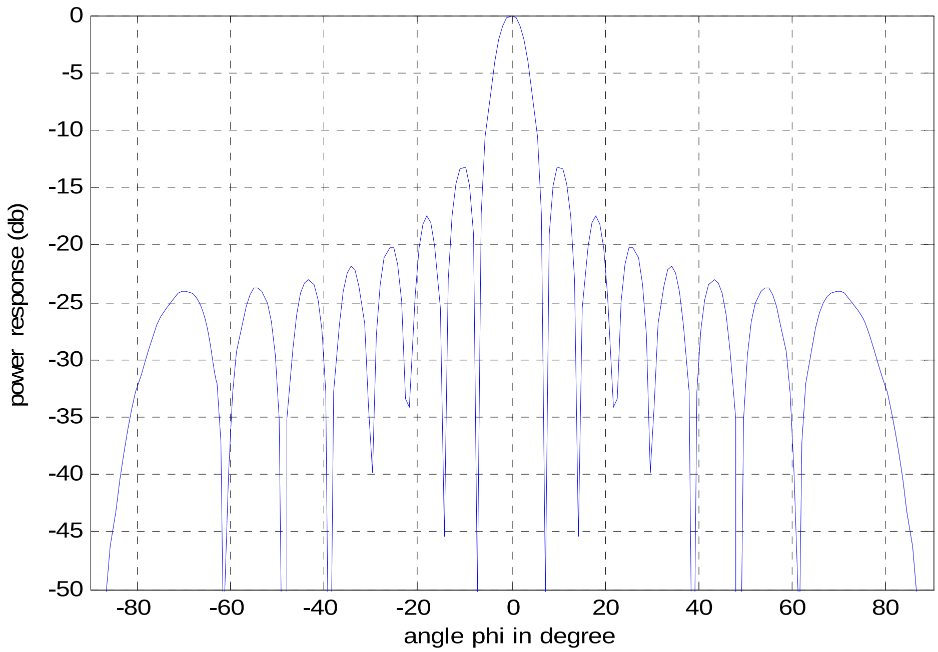

Fig6.3. A sample beam-pattern for a spatial matched filter for an N=16 element ULA and $\phi = 0^{\circ}$

6.3 Effect of the element spacing on the beam pattern

We determined that the element spacing must be $d \leq \lambda/2$ to prevent spatial aliasing [2]. Here we relax this restriction and look at various element spacing and resulting array characteristics, namely, their beam-pattern. In fig.6.4 we show the beam-pattern of spatial matched filter with $\phi = 0^{\circ}$ for ULAs with element spacing of λ/4, λ/2, λ, and 2λ (equal size aperture of 10λ with 40, 20, 10, and 5 elements, respectively). We note that the beam-pattern for λ/4 and λ/2 are identical with equal-sized main-lobes and the first side-lobe having a height of -13dB. In the case of under-sampled array (d= λ and d=2λ), we see the same structure (beam-width) around the look direction but also note the additional peaks in the in the beam-pattern (0 dB) at ± 90° for d= λ and even closer for d=2λ. These additional lobes in the beam-pattern are known as grating lobes. Grating lobes create spatial ambiguities: that is, signal incident on the array from the angle associated with grating lobes look just like signals from the direction of interest.

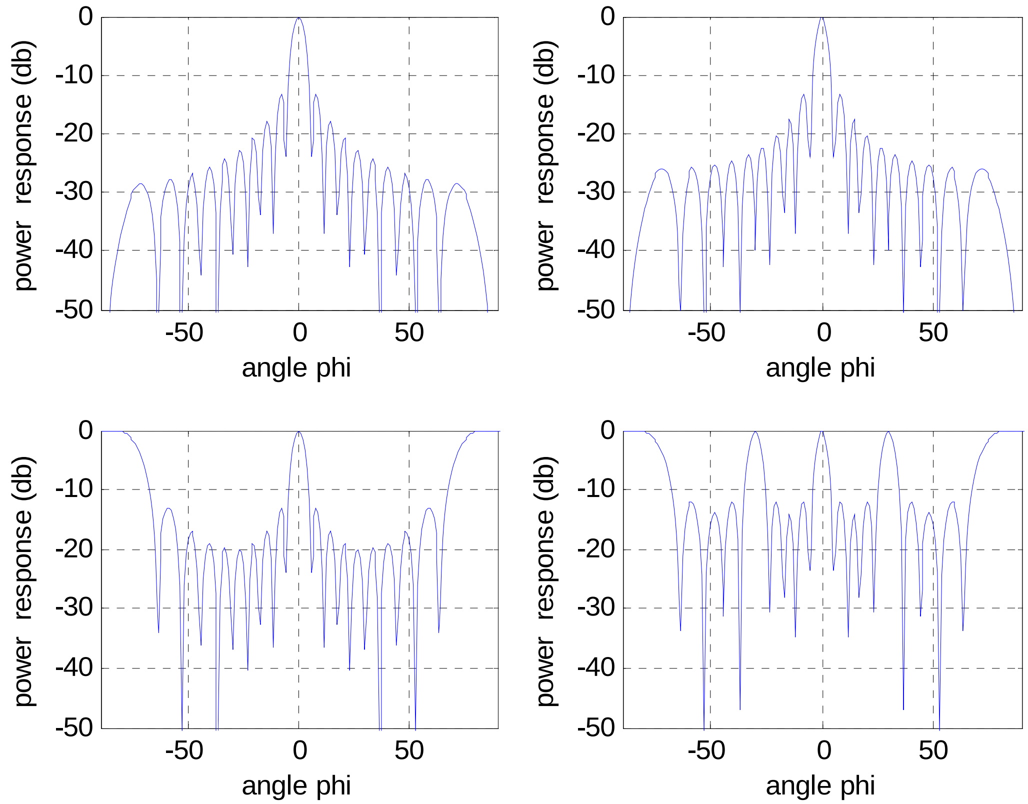

Fig6.4. Beam-pattern for different element spacing λ/4, λ/2, λ, and 2λ respectively (equal size aperture of 10λ with 40, 20, 10, and 5 elements respectively).

6.4 Effect of the array aperture on the beam pattern

The aperture [2] is the finite area over which a sensor collects spatial energy. In the case of ULAs, the aperture is the distance between the first and last elements. In general the designer of array yearns for a much aperture as possible. The greater the aperture, the finer the resolution of the array, which is its ability to distinguish between closely spaced sources. We illustrate the effect of aperture on resolution, using few representative beam-patterns. Fig.6.5 show beam-pattern for N=4, 8, 16, and 32 with enter elements spacing fixed at d= λ/2 (non-aliasing condition). Therefore, the corresponding aperture in wavelengths is 2λ, 4λ, 8λ, and 16λ. Clearly, increasing the aperture yields better resolution, with factor-of-2 improvement for each of the successive twofold increase in aperture length. The label of first side lobe is always -13dBbelow the main lobe peak.

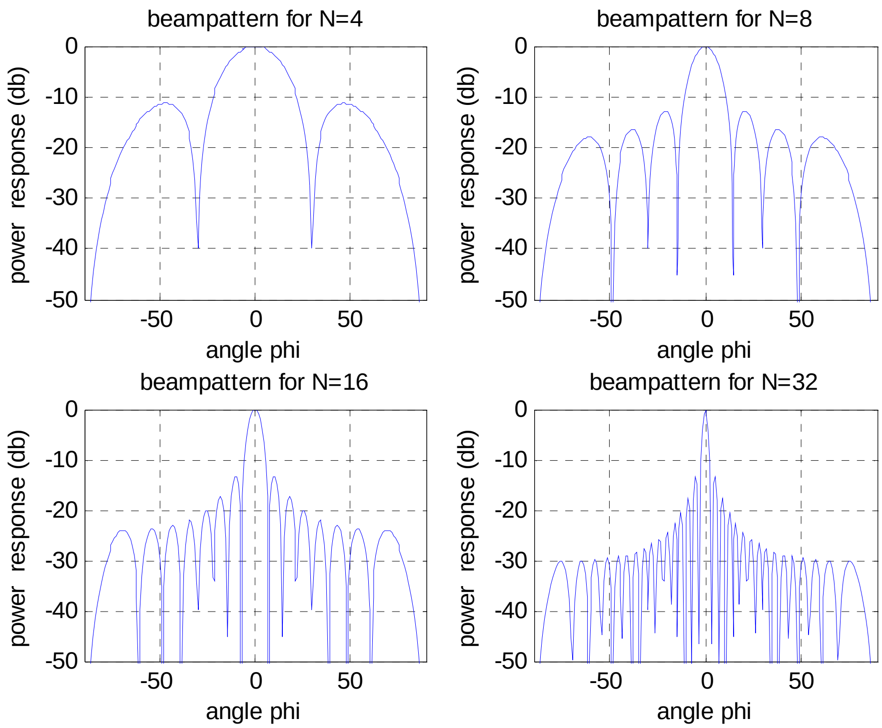

Fig.6.5. Beam-pattern for different aperture sizes2λ, 4λ, 8λ, and 16λ respectively (common element spacing of d= λ/2 with 4, 8, 16, and 32 elements respectively.

6.5 Adaptive Beam-forming Algorithm

The adaptive algorithm used in the signal processing has a profound effect on the performance of a Smart Antenna system. Although the smart antenna system is sometimes called the "Space Division Multiple Access", it is not the antenna that is smart. The function of an antenna is to convert electrical signals into electromagnetic waves or vice versa but nothing else. The adaptive algorithm is the one that gives a smart antenna system its intelligence. Without an adaptive algorithm, the original signals can no longer be extracted.

Different adaptive algorithms were developed for different purposes and tasks. The task of the algorithm in a Smart antenna system is to adjust the received signals so that the desired signals are extracted once the signals are combined. Various methods can be used in the implementation of an adaptive algorithm. In this project, the adaptive algorithm is implemented in MATLAB code.

In comparison, the hearing system of a human being is much like a smart antenna system. Like the antenna, our ears pick up all sound waves from the surrounding environment. From what has been received, the human brain picks out the important information. For example, people are able to listen to a conversation even though the conversation may take place in a very noisy environment. The desired signal can be mixed with other interference like traffic noise, background music, etc., but the human brain is able to suppress the unrelated sounds and concentrate on the conversation. Furthermore, a human can even listen to sound which is weaker than the interference. The adaptive algorithm in a smart antenna system serves a similar purpose as the brain in this analogy, however it is less sophisticated. Our brain can perform the above signal selection and suppression with only two ears, but multiple antennas are required for the adaptive algorithm so that enough information on the user signals can be acquired to perform the task. In human beings, some people are more intelligent than others. In order for them to be more intelligent, they have to have a more developed brain. Similarly, some algorithms are smarter than other algorithms. A smart algorithm usually requires more resources than algorithms that are less intelligent. Unlike our brain which is a free resource, more resources in the world of technology always mean more expensive components and more complicated system.

In this project, we look at the two type of method: block adaptive and sample-by-sample method. Block implementation of the adaptive beam-former uses a block of data to estimate the adaptive beam-forming weight vector and is known as "sample matrix inversion (SMI)". The sample-by-sample method updates the adaptive beam-forming weight vector with each sample. The sample-by-sample method, here we used, are least mean square (LMS) algorithm, constant modulus algorithm (CMA) ,least square CMA and recursive least square (RLS) algorithm. All algorithms are described in more details in the following sections.

6.5.1 Sample Matrix Inversion

For a ***N***-element antenna array, the baseband received signal vector X is given by

$$\underline{x}(t) = \sum_{i=1}^{M} s_i(k)\underline{a}_i(\theta_i) + \underline{n}(k) \quad (6.4)$$

Where $\underline{x}(k) = [x_1(k)x_2(k)..........x_N(k)]$ is $1 \times N$ complex valued vector and ***k*** denotes discrete time. The co-channel transmitted signals are represented by $s_i(k)$, for $i = 1,2,........M$. The $1 \times N$ row vector $\underline{a}_i$ is the array response vector associated with the i^{th} transmitted signal, which models the antenna array gain and phase across each of the elements. This is a function of angle-of-arrival, θ_i of the received signal. Noise is modeled by $\underline{n}(k) = [n_1(k)n_2(k)......n_N(k)]$, a $1 \times N$

vector of complex white noise with variance N_0^2 . The assumption is that each of the transmitted signals and noise sequences are mutually uncorrelated. The sensor outputs are each multiplied by a complex weight $w_i(k)$ which may vary with time, and then summed to produce the output $y(k)$. The goal is to adjust the complex weights w_i to improve reception of the signal of interest (SOI). The array output is expressed as

$$y(k) = \sum_{i=1}^{N} w_i(k)x_i(k) = \underline{x}(k)\underline{w}(k) \tag{6.5}$$

Where $\underline{w}(k)$ is the $N \times 1$ column vector of beam-former weights. The weight vector that minimizes the mean squared error is given by

$$\underline{w}_{opt} = R^{-1}P \tag{6.6}$$

Where R=E[$\underline{x}^H(k)\underline{x}(k)$]and P=E[$\underline{x}^H(k)d(k)$]. The Sample Matrix inversion (SMI) [2, 3] method is a technique used to approximate the solution to the minimum mean square error (MMSE) problem. It assumes that there is a known training sequence d (k) which occurs in the SOI data, that $s_j(k) = d(k)$ for some j, k.

First, K samples of the signal vector X are collected in a K x N matrix

$$X_K(k) = \begin{bmatrix} x_1(k) & \ldots x_N(k) \\ x_1(k+1) & x_1(k+1) \\ . & . \\ . & . \\ x_1(k+K-1) & \ldots x_N(k+K-1) \end{bmatrix} \tag{6.7}$$

This sample is used to form an estimate of the $N \times N$ covariance matrix

$$\hat{R}(k) = X_K^H(k)X_K(k) \tag{6.8}$$

and $N \times 1$ cross-covariance vector

$$\hat{P}(k) = X_K^H(k)\underline{d}(k) \tag{6.9}$$

where, $\underline{d}(k) = [d(k)d(k+1)......d(k+K-1)]^T$ is a $K \times 1$ column vector. The approximation to the solution of MMSE problem is calculated as:

$$\underline{\hat{w}}(k) = \hat{R}(k)^{-1}\hat{P}(k) \tag{6.10}$$

SMI adaptation results in poor interference cancellation performance. This is due to the inadequate estimate of R and P using a finite size K block of array data. Also, in an environment where signals are not continuously transmitted, the problem of partial burst overlap of interferers further degrades SMI performance.

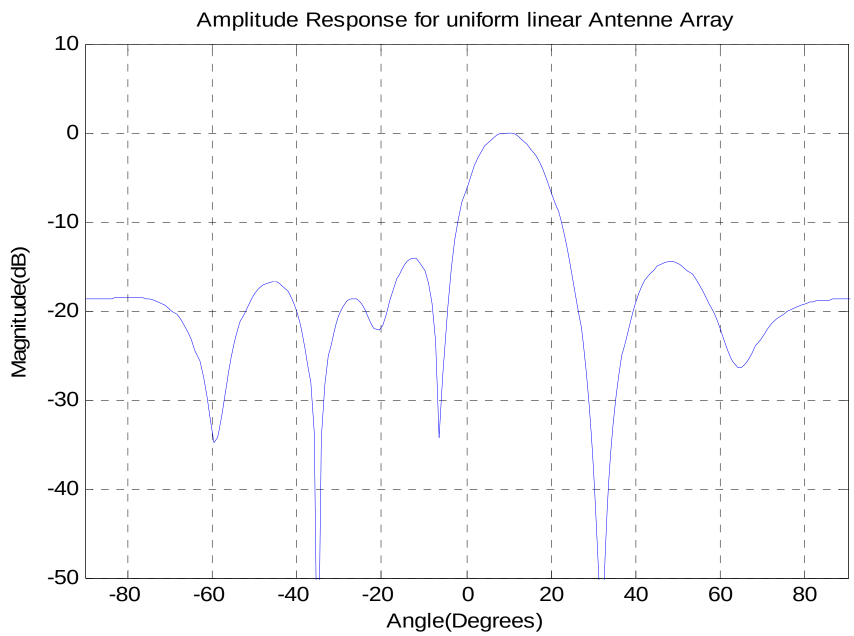

Fig.6.6 the beam-pattern of SMI adaptive beam-former for 8-elements ULA. The signal of interest and two interfering signal are arriving at 10°, -35° and 32° respectively.

6.5.2 Least Mean Square Algorithm

This algorithm was first developed by Widrow and Hoff in 1960 [2, 10, 11]. The design of this algorithm was stimulated by the Wiener-Hopf equation. By modifying the set of Wiener-Hopf equations with the stochastic gradient approach, a simple adaptive algorithm that can be updated recursively was developed. This algorithm was later on known as the least-mean-square (LMS) algorithm.

The algorithm contains three steps in each recursion: the computation of the processed signal with the current set of weights, the generation of the error between the processed signal and the desired signal, and the adjustment of the weights with the new error information. The following equations summarize the above three steps.

$$\hat{d}(n) = w_1^*(n)u_1(n) + w_2^*(n)u_2(n) + \ldots + w_t^*(n)u_t(n) \tag{6.11}$$

or in matrix form,

$$\hat{d}(n) = \boldsymbol{w}^H(n)\boldsymbol{u}(n) \tag{6.12}$$

$$e(n) = d(n) - \hat{d}(n) \tag{6.13}$$

$$\boldsymbol{w}(n+1) = \boldsymbol{w}(n) + \mu\boldsymbol{u}(n)e^*(n) \tag{6.14}$$

The ***w*** in the above equations is a vector which contains the whole set of weights. The *H* in Eqn.(6.12) represents the Hermitian transpose of a vector, i.e. the vector is transposed and each element in the vector is replaced by its conjugate. Here, we have taken eight elements, so there are eight *u's* for each symbol received at time *n*. All eight weights are updated according to Eqn.(6.14) in each recursion. At time zero, all weights are initialized to have a value of zero. The symbol μ in Eqn.(6.14) is called the step size parameter. The value of this parameter affects the settling time and the steady state error of the LMS algorithm. A large step-size allows fast settling but causes poor steady state performance. On the other hand, a small step-size decreases the steady state error but compromises the rate of convergence. The current value of this parameter is selected by trying out different values in the algorithm. The simulation result for LMS algorithm is as following:

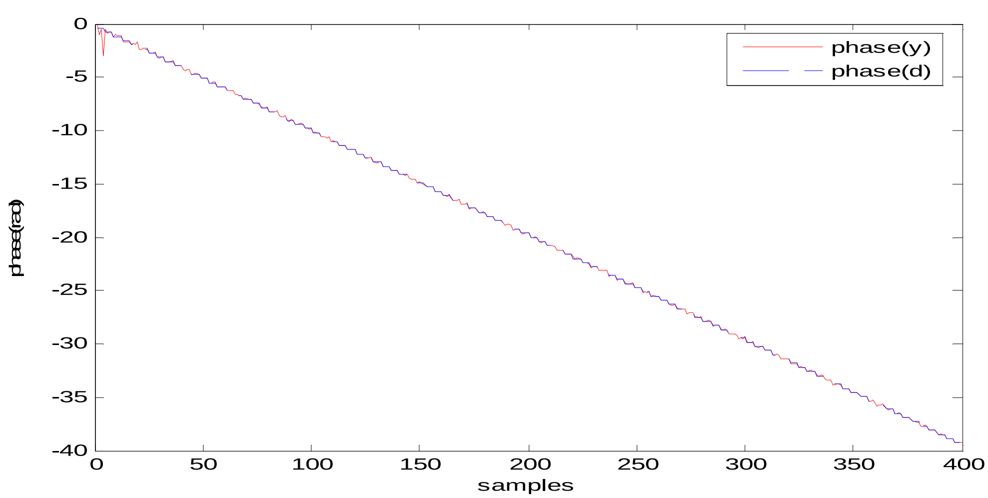

Fig.6.7. Comparison of the phase of desired signal and output for LMS

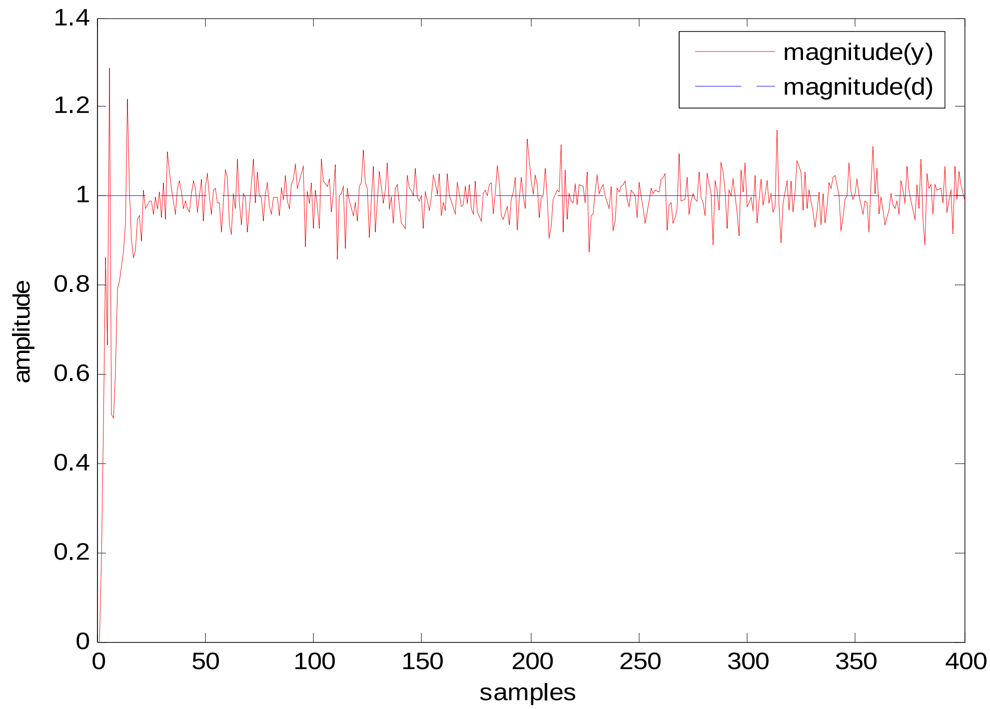

Fig.6.8. Comparison of the amplitude of desired signal and output for LMS

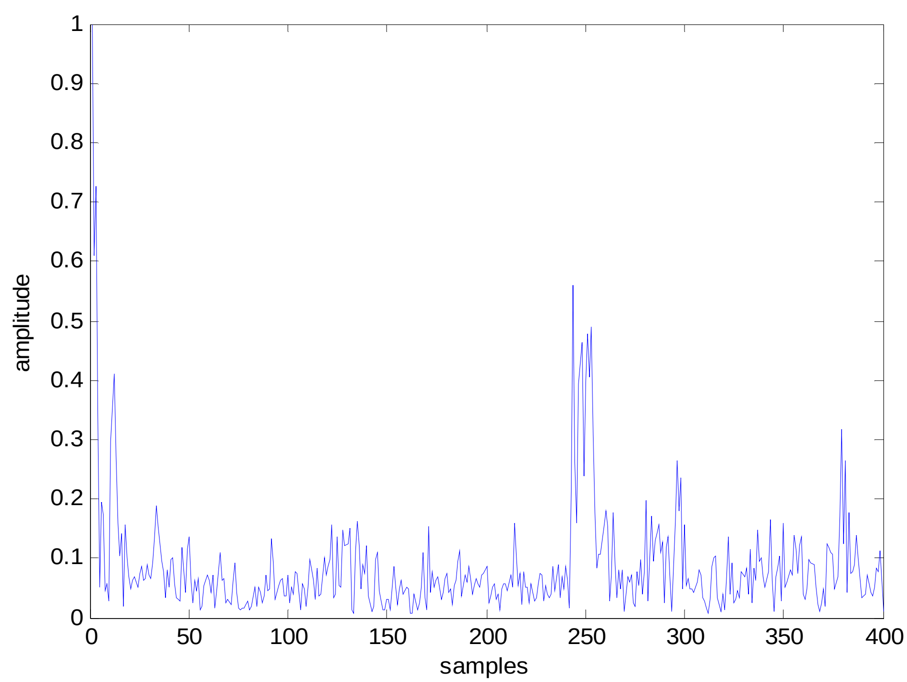

Fig.6.9. Performance curve for LMS algorithm

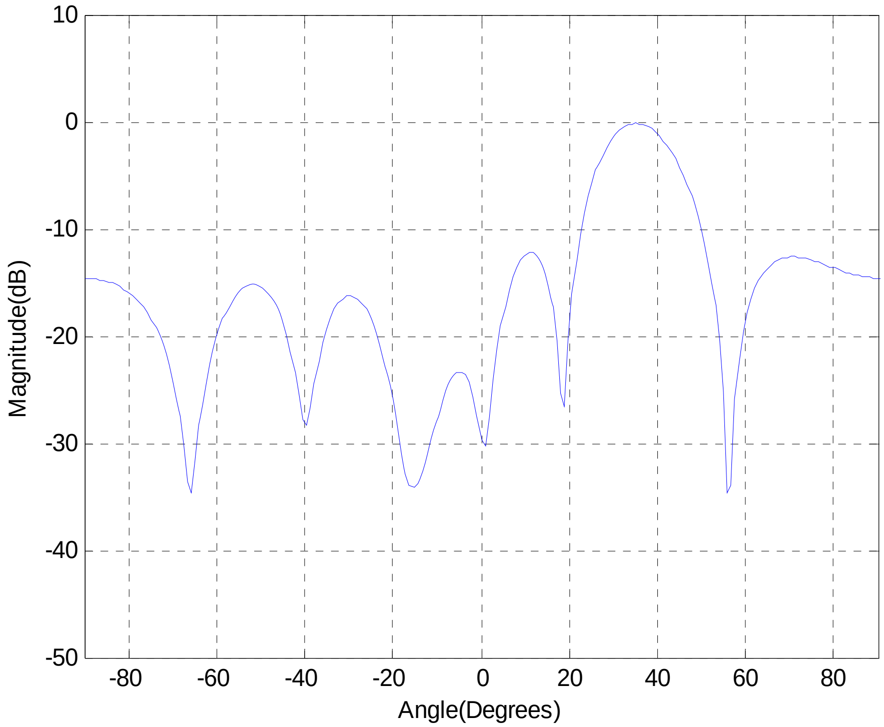

Fig.6.10. The beam-pattern of LMS algorithm adaptive beam-former for 8-elements ULA. The signal of interest and two interfering signal are arriving at 35, 0 and -20 degree respectively.

6.5.3 Constant Modulus Algorithm

Many adaptive beam forming algorithms are based on minimizing the error between reference signal and array output. The reference signal is typically a training sequence used to train the adaptive array or a desired signal based upon a priori knowledge of nature of the arriving signals. In the case where a reference signal is not available one must resort to an assortment of optimization techniques that are blind to exact content of the incoming signals.

The constant modulus algorithm (CMA) [10 ,15] is used for blind equalization of signals that have a constant modulus .The minimum shift key (MSK) signal, for example, is a signal that has the property of a constant modulus.

The constant modulus algorithm is Godard Algorithm. Godard was the first to propose a family of constant modulus blind equalization algorithms .The algorithm contains three steps in each recursion: the computation of the processed signal with the current set of weights(Initial weight w(1) are chosen), the generation of the error , and the adjustment of the weights with the new error information. The following equations summarize the above three steps.

$$y(k) = W^H . X(k) \tag{6.15}$$

$$e(k) = (y(k) - \frac{y(k)}{|y(k)|}) \tag{6.16}$$

$$W(k+1) = W(k) + \mu e(k) . X(k) \tag{6.17}$$

In equation (6.15), W represents weight vector of the arrays. Here, eight elements have been taken, so each weight vector contains eight elements of weight. In equation (6.16), e(k) is error at k iteration. The symbol μ in Eqn.(6.17) is called the step size parameter. The value of this parameter affects the settling time and the steady state error of the CM algorithm. The simulation results for constant modulus algorithm are as following:

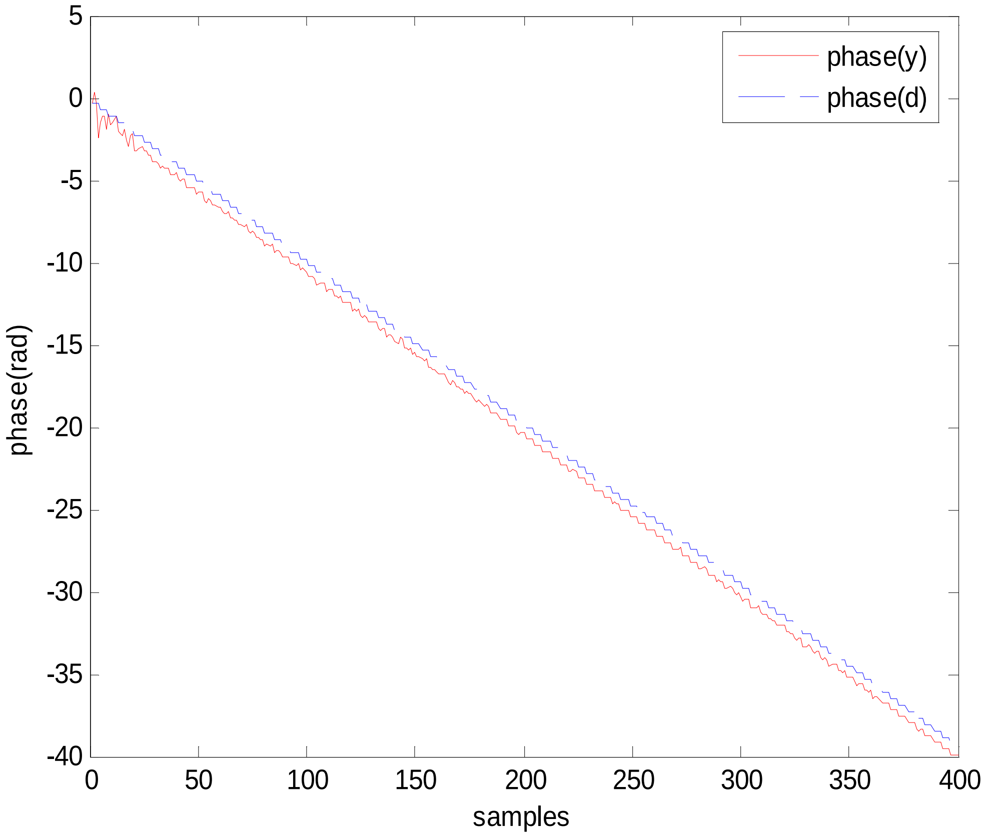

Fig6.11. Comparison of the phase of desired signal and array output for CMA

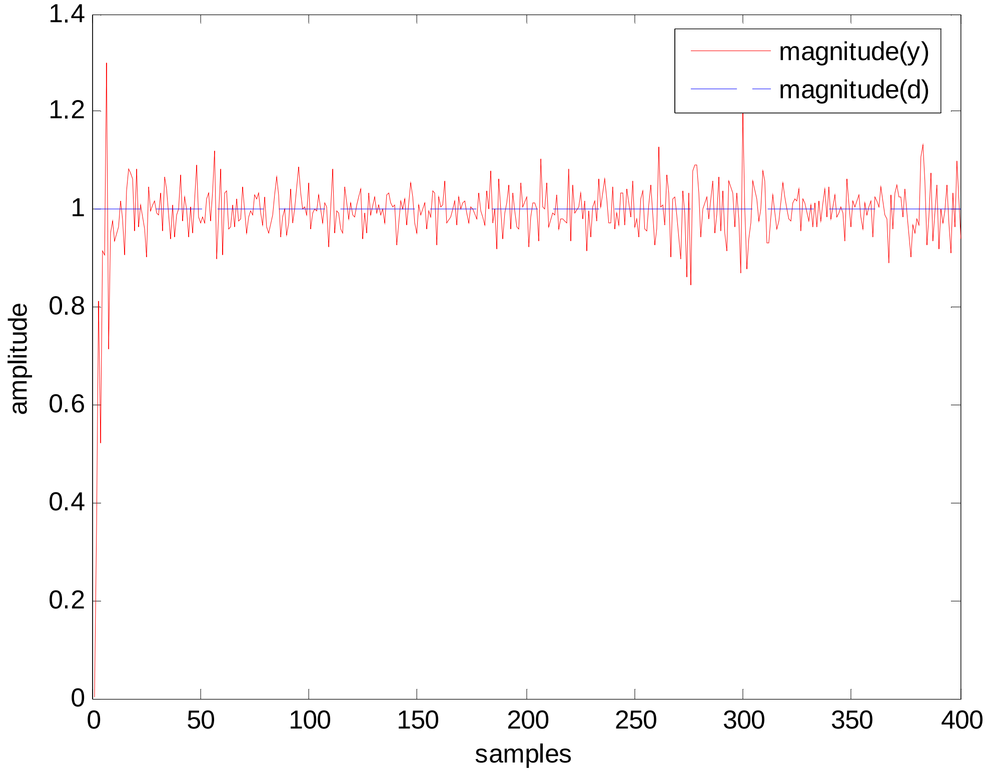

Fig.6.12.Comparison of the amplitude of desired signal and array output for CMA

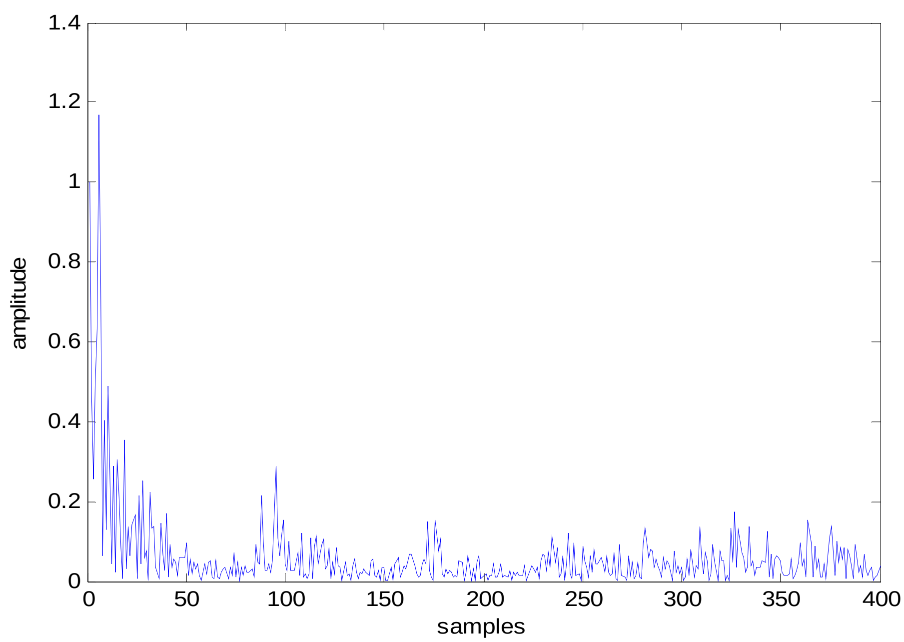

Fig.6.13. Performance curve for CMA

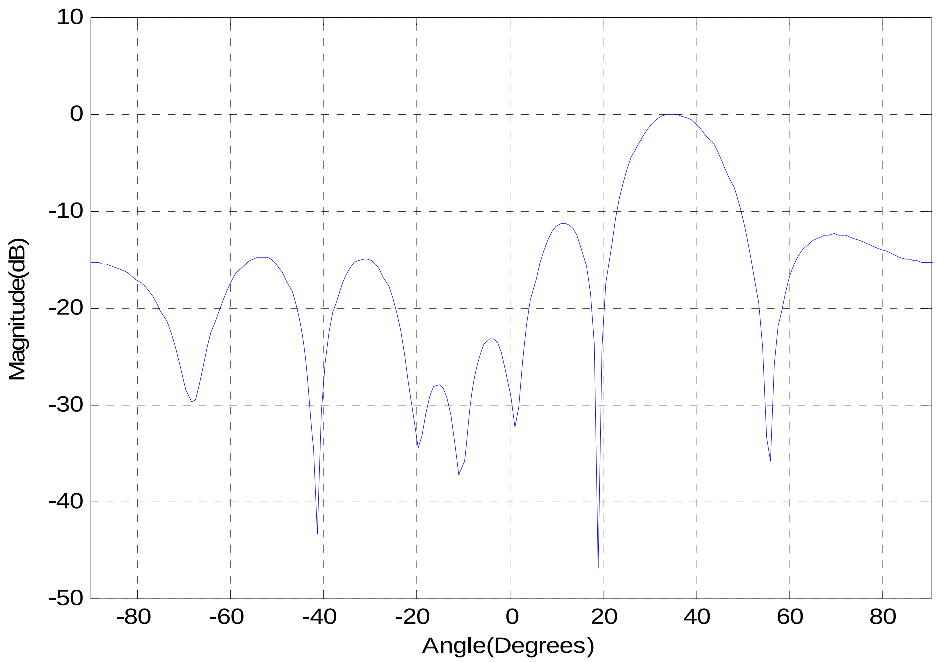

Fig.6.14. The beam-pattern of CMA adaptive beam-former for 8-elements ULA. The signal of interest and two interfering signal are arriving at 35, 0 and -20 degree respectively.

6.5.4. Least Square Constant Modulus

One severe disadvantage of the Godard CMA is slow convergence time .The slow converges limits the usefulness of the algorithm in the dynamic environment where the signal must be captured quickly. This also limits the usefulness of CMA when channel conditions are rapidly changing. The previous Godrad CMA is based upon the method of steepest descent by taking the gradient of the cost function. A faster algorithm was developed by Agee[16] using the method of non-linear least square. The least square algorithm is also known as the Gauss method based upon the work of Gauss in 1795.This method is known as least square constant modulus algorithm.

The least-squares constant modulus algorithm (LSCMA) is summarized as following:

$$\begin{aligned} \mathbf{w}_{k+1} &= \mathbf{w}_k - (\mathbf{X}^* \mathbf{X}^T)^{-1} \mathbf{X}^* (\mathbf{y}_k - \delta_k) \\ &= (\mathbf{X}^* \mathbf{X}^T)^{-1} \mathbf{X}^* \delta_k \end{aligned} \qquad (6.18)$$

Where X is the input data matrix and y_k and δ_k, are the output-data and complex-limited output-data vectors,

$$\mathbf{X} = [\mathbf{x}(1), \mathbf{x}(2), \cdots \mathbf{x}(N)] \tag{6.19}$$

$$\mathbf{y}_k = [y_k(n)] = \mathbf{X}^T \mathbf{w}_k \tag{6.20}$$

$$\delta_k = [\delta_k(n)] = [y_k(n)/ \mid y_k(n) \mid] \tag{6.21}$$

While only one block of data is used to implement the LS-CMA algorithm iterates through n values until convergence. The initial weight vector W (1) are chosen, the complex-limited output data vector $\delta_k(1)$ is calculated, and then the next weight vector W (2) is calculated, and the iteration continue until satisfactory convergence is satisfied. This is called the static LS-CMA because only one block, of length K, is used for the iteration process. The least square constant modulus algorithm bears resemblance to the SMI algorithm. The simulation results for LS-CMA are as following:

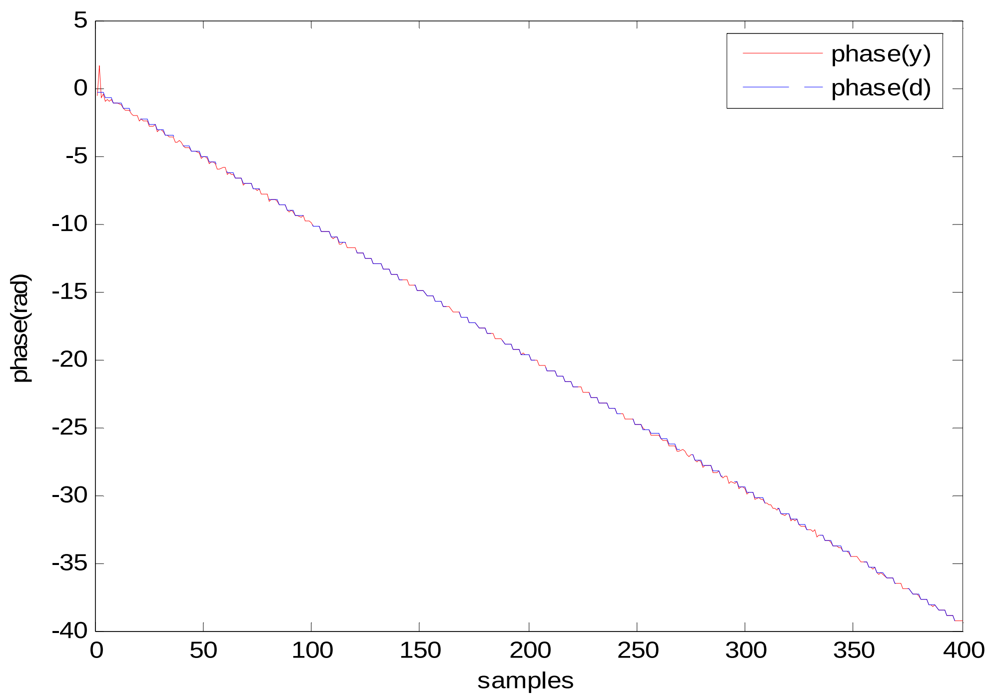

Fig.6.15. Comparison of the phase of desired signal and array output for LS-CMA

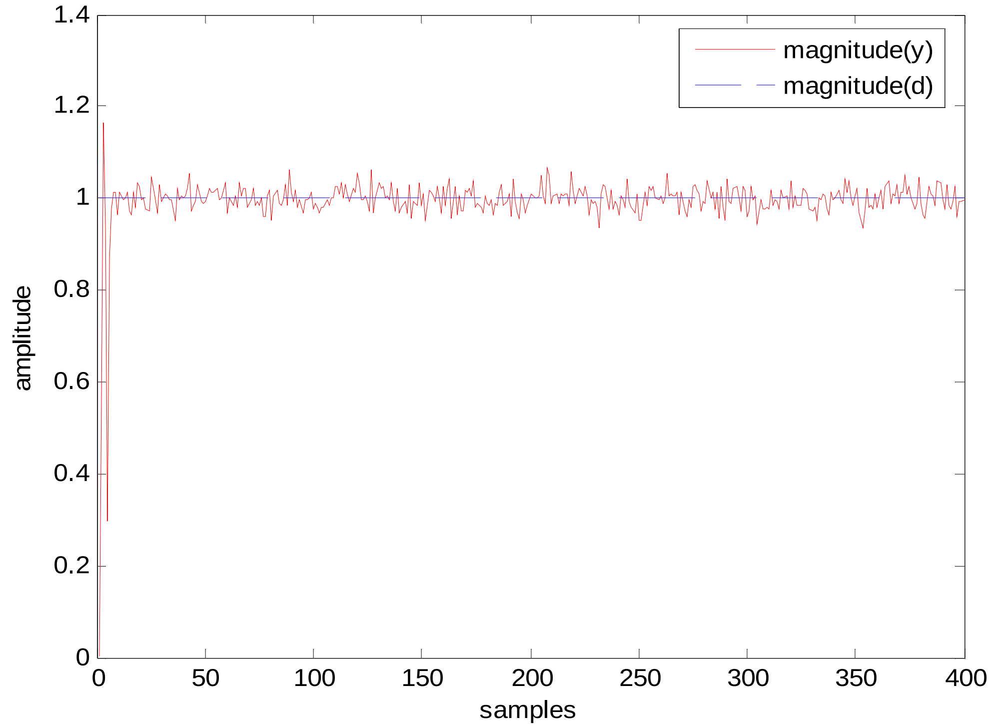

Fig 6.16 Comparison of the amplitude of desired signal and array output for LS-CMA.

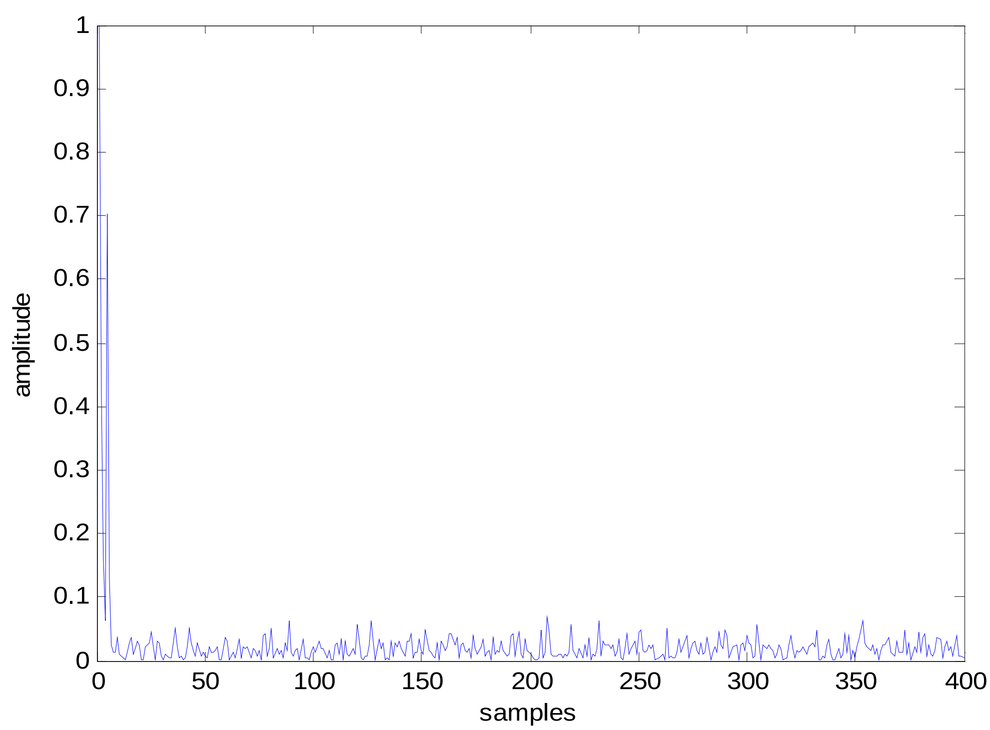

Fig6.17. Performance curve for LS-CMA algorithm.

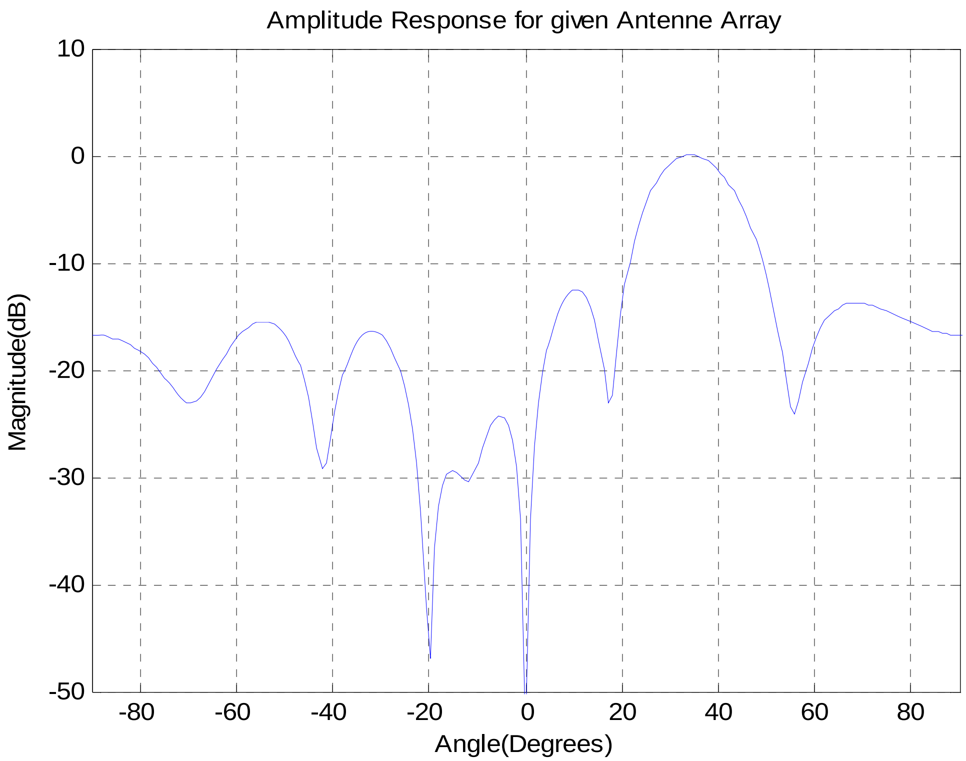

Fig.6.18 The beam-pattern of LS-CMA algorithm adaptive beam-former for 8-elements ULA. The signal of interest and two interfering signal are arriving at 35, 0 and -20 degree respectively.

6.5.5 Recursive Least Square Algorithm

The recursive least-squares (RLS) algorithm [10] uses a different approach in carrying out the adaptation. Instead of minimizing the mean square error as in the LMS algorithm, the sum of the squared errors of different set of inputs is the subject of minimization. This algorithm was first derived from the Kalman filter. Although it is intended to be used in a multi-tap transversal filter where the squared error information is sampled over a varying time frame, this method also works in our system where input information originates from different elements. The following equations illustrate the update procedures in each recursion with the RLS algorithm.

$$\boldsymbol{k}(n) = \frac{\lambda^{-1}\boldsymbol{P}(n-1)\boldsymbol{u}(n)}{1+\lambda^{-1}\boldsymbol{u}^{H}(n)\boldsymbol{P}(n-1)\boldsymbol{u}(n)} \tag{6.22}$$

$$\xi(n) = d(n) - \boldsymbol{w}^H(n-1)\boldsymbol{u}(n) \tag{6.23}$$

$$\boldsymbol{w}(n) = \boldsymbol{w}(n-1) + \boldsymbol{k}(n)\xi^*(n) \tag{6.24}$$

$$\boldsymbol{P}(n) = \lambda^{-1}\boldsymbol{P}(n-1) - \lambda^{-1}\boldsymbol{k}(n)\boldsymbol{u}^H(n)\boldsymbol{P}(n-1) \tag{6.25}$$

The ***P*** in the above equations is first initialized to $\delta^{-1}I$ where δ is a small positive constant and ***I*** is an identity matrix. All ***w***'s are again initialized to zeroes. ***k*** is a vector called the Kalman gain factor. λ is the forgetting factor and is supposed to weight the error value differently depending on the ages of the received signals in a transversal filter. Since the filter in this system does not have a transversal architecture, it makes more sense to leave it as 1. The simulation result for RLS algorithm is as following:

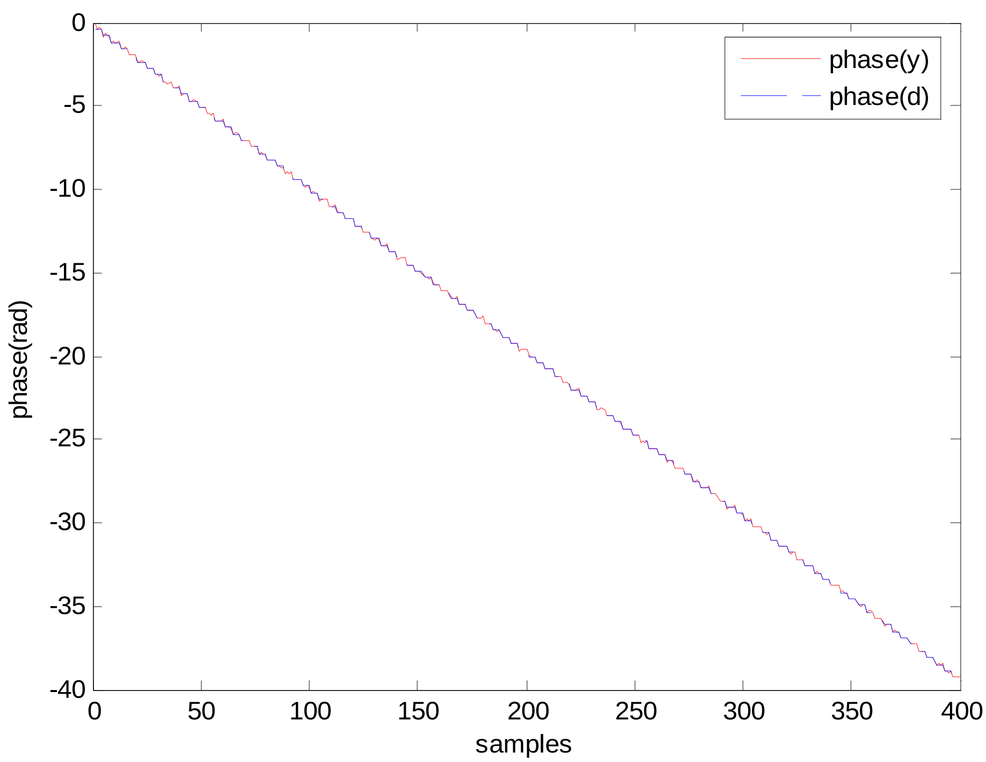

Fig.6.19. Comparison of the phase of desired signal and array output for RLS.

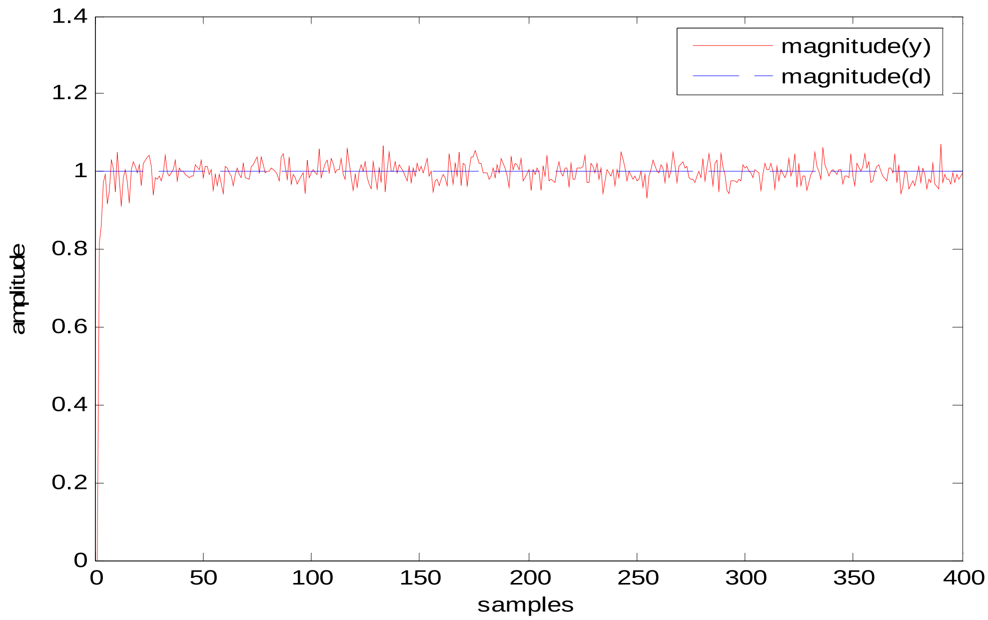

Fig.6.20. Comparison of the amplitude of desired signal and array output for RLS.

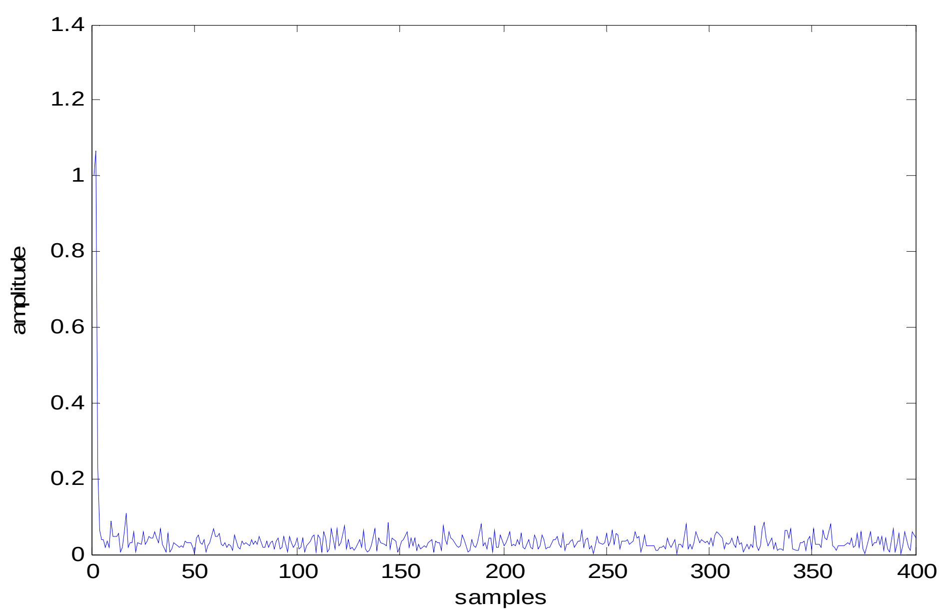

Fig6.21. Performance curve for RLS algorithm

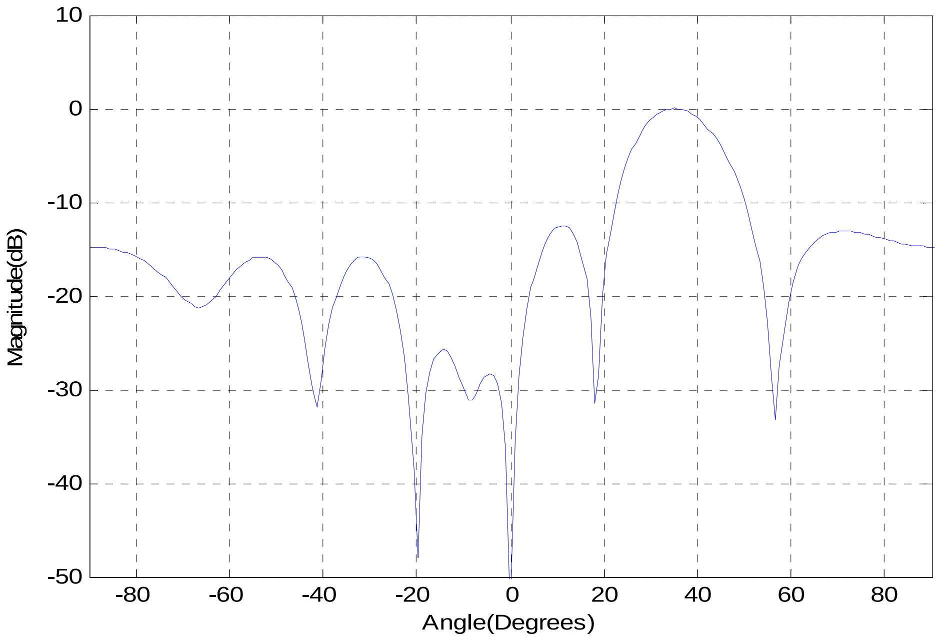

Fig.6.22. The beam-pattern of RLS algorithm adaptive beam-former for 8-elements ULA . The signal of interest and two interfering signal are arriving at 35, 0 and -20 degree respectively.

CHAPTER 7

COMPARISON OF BEAMFORMING ALGORITHMS

7.1 Comparison of Algorithms

In the simulation, the smart antenna of 8-elements has been taken. The signal arrives at 10°.Two interfering signals are at -35° and 32°. The smart antenna algorithms compute the antenna weights for all eight antenna elements so that the signal-to-noise-and-interference ratio (SINR) becomes optimum.

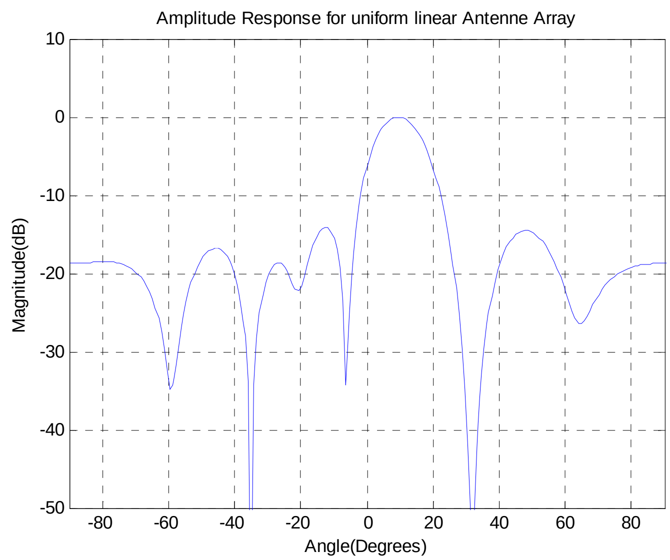

Fig7.1 SMI adaptive beam-forming for 8-elements ULA. The signal of interest and two interfering signal are arriving at 10°, -35° and 32° respectively.

The SMI adaptive beam-former uses a block of code to get optimum weight; therefore it is not suitable for non stationary environment. Since the sample-by-sample adaptive beam-former alter its weight with each new sample, it can dynamically update its response for such a changing scenario.

Another important distinction between sample and block adaptive method is the inclusion of the signal of interest in each sample and thus in the correlation matrix. Therefore, for sample adaptive method, we can not use signal free version of the correlation matrix, that is, the interference plus noise correlation matrix, but rather must use the whole correlation matrix .The inclusion of the signal in the correlation matrix has profound effect on the robustness of the adaptive beam-former in the case of signal mismatch.

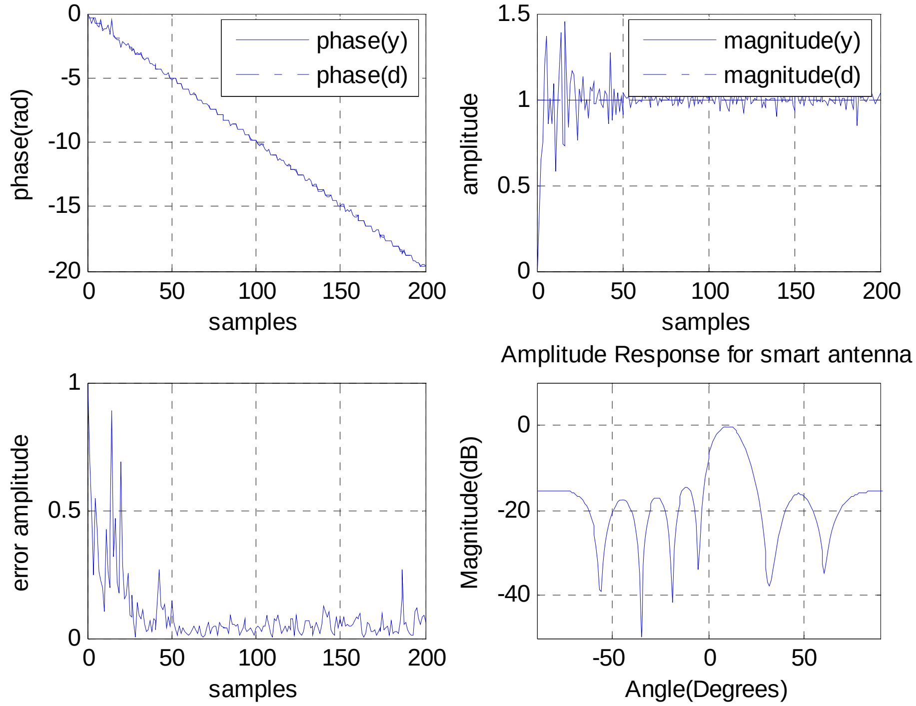

Fig7.2. LMS adaptive beam-forming for 8-elements ULA and μ =0.05. The signal of interest and two interfering signal are arriving at 10°, -35° and 32° respectively.

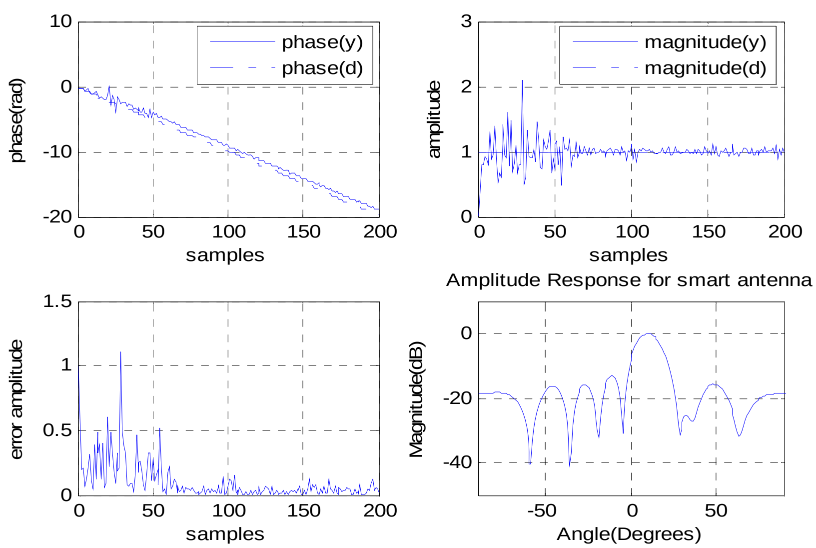

Fig7.3. CMA adaptive beam-forming for 8-elements ULA and μ =0.05. . The signal of interest and two interfering signal are arriving at 10°, -35° and 32° respectively.

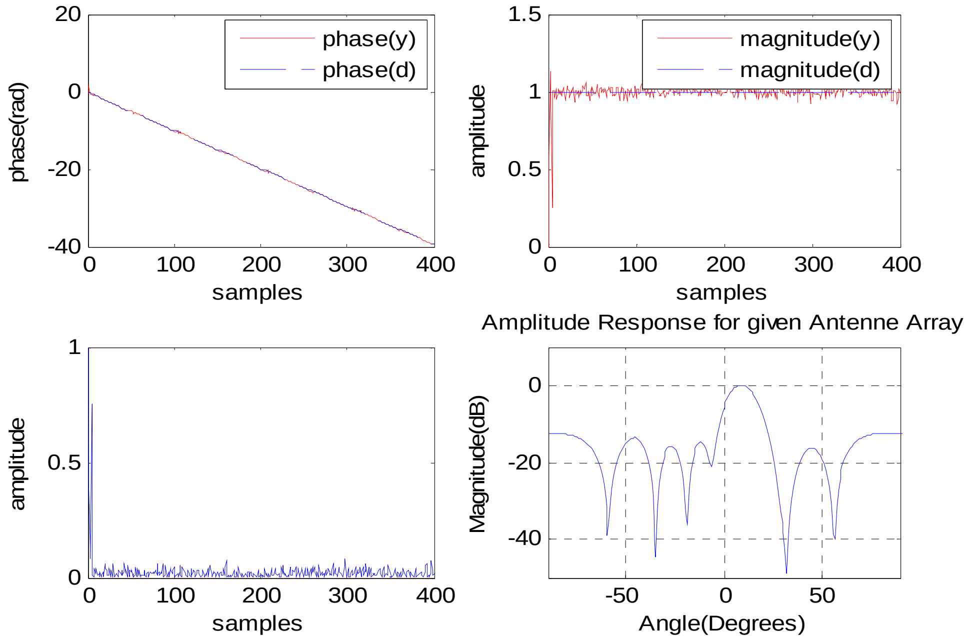

Fig.7.4. LS- CMA adaptive beam-forming for 8-elements ULA and μ =0.05. . The signal of interest and two interfering signal are arriving at 10°, -35° and 32° respectively

The constant modulus algorithm converges slower then least square algorithm .During the effort to simulate constant modulus algorithm it was clear that the algorithm is less stable than the least mean square algorithm .The constant modulus algorithm seems to be more sensitive to gradient constant μ .

The simulation results support the stability and convergence results presented in fig.7.3, fig.7.4, and demonstrate the superior adaptivity of the least-squares CMA over the steepest-descent CMA. One severe disadvantage of the CMA is slow convergence time .The slow converges limits the usefulness of the algorithm in the dynamic environment where the signal must be captured quickly. This also limits the usefulness of CMA when channel conditions are rapidly changing. The CMA is based upon the method of steepest descent by taking the gradient of the cost function. A faster algorithm was developed using the method of non-linear least square.

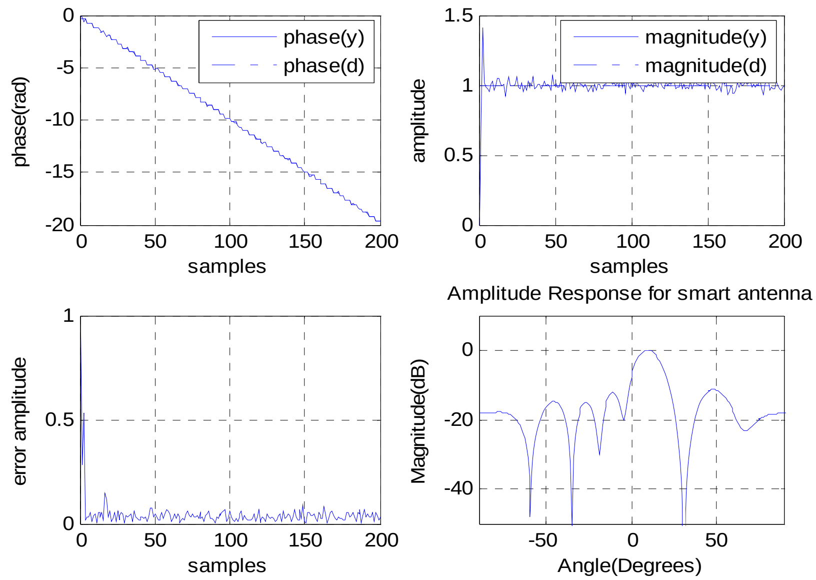

Fig7.5. RLS adaptive beam-forming for 8-elements ULA, $\lambda = 1$ and $\delta = 0.004$. The signal of interest and two interfering signal are arriving at 10°, -35° and 32° respectively.

An important feature of the recursive least square algorithm is that its rate convergence is typically an order of magnitude faster then that of the simple least square algorithm ,due to fact that recursive least square algorithm whiten the input data by using the inverse correlation matrix of the data, assumed to be zero mean . This improvement, however, achieved at the expense of an increase in computational complexity of the recursive least square algorithm. We verified with simulation result that the smart antenna minimizes interference noise, and RLS algorithm is better than CMA, LS-CMA and LMS algorithm.

CHAPTER 8

CONCLUSION AND SCOPE OF FURURE WORK

8.1 Conclusion

The principle reason for the growing interest in smart antenna systems is the capacity increase and low power consumption. In densely populated areas, mobile systems are normally interference-limited, meaning that the interference from other users is the main source of noise in the system. This means that the signal-to-interference ratio (SIR) is much larger than the signal-to-noise ratio (SNR). In general, smart antennas will increase the SIR by simultaneously increasing the useful received signal level and lowering the interference level.

Just by looking at the equations that describe the updating procedures, it is clear that the RLS algorithm is more complicated than the LMS algorithm. The RLS algorithm has a computation complexity in the order of square of M, where M is the number of taps or in this case, the number of elements. For the LMS algorithm, the complexity is in the order of M. However, the RLS algorithm has much better performance than LMS algorithm. Its rate of convergence is consistent and is independent of the eigen values of the received signals. It only takes approximately 2M + 1 iterations to get to the steady state. Moreover, the steady state error is smaller in magnitude than the one obtained by the LMS algorithm.

CMA algorithm has slower convergence then LMS algorithm. The constant modulus algorithm is unsupervised algorithm. It optimizes weight of elements in the array without reference signal. The reference signal is typically a training sequence used to train the adaptive array or a desired signal based upon priory knowledge of nature of the arriving signal. The LS-CMA is also an unsupervised algorithm, but its performance is better then CMA algorithm. The chief advantage of the LS-CMA is that it can converge up to 100 times faster then the conventional CMA algorithm.

Wireless operators face a sizeable technology challenge as they pursue growth through data and multimedia services. We have outlined how smart antenna system will contribute to meeting that challenge, along with continued spectrum efforts and new scale economies in the wireless supply base. We can expect, smart antenna system will play a key role in wireless communication.

8.2 Scope of future work

The LS-CMA, also known as static LS-CMA, computed the weights simply based upon a fixed block of sampled data. In order to maintain up-to-date adaptation in dynamic signal environment, it is better to update the data block for each iteration Thus a dynamic LS-CMA algorithm [15] is more appropriate. The dynamic LS-CMA is a modification of the previous static version.

The neural network approach [17] is used to the problem of finding the weights of one- (1-D) and two-dimensional (2-D) adaptive arrays. In modern cellular satellite mobile communications systems and in global positioning systems (GPS's), both desired and interfering signals change their directions continuously. Therefore, a fast tracking system is needed to constantly track the users and then adapt the radiation pattern of the antenna to direct multiple narrow beams to desired users and nulls interfering sources. In the approach suggested, the computation of the optimum weights is accomplished using three-layer radial basis function neural networks (RBFNN). The results obtained from this network are in excellent agreement with the Wiener solution.

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