COMPARATIVE STUDY OF THE ESTIMATION TECHNIQUES FOR DIRECTION OF SIGNAL ARRIVAL IN SMART ANTENNAS

By

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List of Abbreviations

RF	Radio frequency
SDMA	Space division multiple access
FDMA	Frequency division multiple access
TDMA	Time division multiple access
CDMA	Code division multiple access
LAN	Local area network
DSP	Digital signal processor
TDD	Time division duplex
FDD	frequency division duplex
DOA	direction of arrival
ESPRIT	estimation of signal parameter via rotational invariance technique
CLOSEST	Close to the steering estimation
FINE	first principle vector
LMS	least mean square
LP	linear prediction
LS	least square
MAP	maximum a posteriori
MDL	minimum description length
ME	maximum entropy
ML	maximum likelihood
MLM	maximum likelihood method
MSE	mean square error
MVDR	minimum variance distortion less response
MUSIC	multiple signal classification
SNR	signal to noise ratio
STD	standard variation
CRLB	Cramers-Rao lower bound
ULA	Uniformly spaced linear array
TAM	Toeplitz approximation method
WSF	weighted subspace fitting
MAP	maximum a posteriori

List of symbols

A	LXM matrix with column being steering vectors
d	incremental spacing of linear equispaced array
F_N	Nyquist frequency
$H(s)$	entropy function
K	No of element in subarray
L	No of elements in array
L_0	No of subarrays
M	No of directional sources
N	No of samples
R	Array correlation matrix
S_θ	steering vector associated with direction θ
$S(f)$	power s
Θ	direction of source
$P_B(\theta)$	Power estimated by Bartlett method as function of θ
$P_{LP}(\theta)$	Power estimated by Linear prediction method as function of θ
$P_{ME}(\theta)$	Power estimated by Maximum entropy method as function of θ
$P_{MN}(\theta)$	Power estimated by Minimum norm method as function of θ
$P_{MU}(\theta)$	Power estimated by MUSIC method as function of θ
$P_{MV}(\theta)$	Power estimated by MVDR method as function of θ

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ABSTRACT

A smart antenna system combines multiple antenna elements with a signal processing capability to optimize its radiation pattern automatically in response to the signal environment. Widespread interest of smart antenna has continued for several decades due to their use in numerous applications. As a whole Smart Antennas examine all aspects of array signal processing, delivering a detailed treatment of antenna array processing schemes, adaptive algorithm to adjust weighting, direction of arrival (DOA) estimation methods, and diversity combining methods that combat fading and reduce error. The objective of this work is to provide a comprehensive study on various techniques concerning estimation of received signals on directions of arrival (DOA).The goal is to achieve maximum SNR, resolution, sensitivity by comparing some techniques. In order to obtain the best possible cancellation of unwanted interferences the gain of the low noise amplifier (LNA) must be adjusted. For optimum processing the typical objective is to maximize the output signal to noise ratio (SNR). For an array with a specified response in the direction of the desired signal, this is achieved by minimizing the mean output power of the processor subject to be specified constraint. In the absence of errors, the beam pattern of the optimized array has the desired response in the signal direction and reduced response in the direction of unwanted interference.

In this research work the three types of DOA estimation method are considered. Firstly conventional method includes spectral estimation, minimum variance distortion less response estimator. Secondly optimal method includes linear prediction, maximum entropy, and maximum likelihood. Finally various Eigen structure methods are also described which includes different versions of multiple signal classification (MUSIC) methods, minimum norm methods, estimation of signal parameters via rotational invariance technique (ESPRIT) method and the weighted subspace fitting method. The power and beam pattern expressions of various techniques have been computed and the output has been analyzed to compare the various estimation methods. It has been found that the structured estimation techniques give the better resolution, better sensitivity than the conventional and optimal methods.

Chapter -1
Smart Antenna System

1.1 Introduction

A smart antenna system combines multiple antenna elements with a signal-processing capability to optimize its radiation and/or reception pattern automatically in response to the signal environment.

An antenna is the port through which radio frequency (RF) energy is coupled from the transmitter to the outside world and, in reverse, to the receiver from the outside world. Yet, the manner in which energy is distributed into and collected from surrounding space has a profound influence on the efficient use of spectrum, the cost of establishing new networks, and the service quality provided by those networks

1.2 Types of Antennas

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

1.2.1Omnidirectional 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. Fig 1.1 shows the radiation pattern of an omnidirectional antenna. 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.

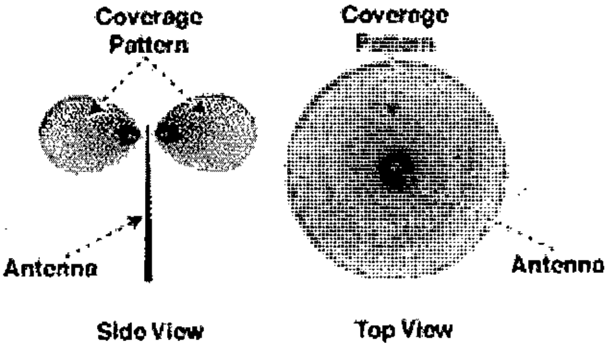

Figure 1.1 Omnidirectional Antenna and Coverage Patterns [1]

Given this limitation, omnidirectional 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), omnidirectional 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 multipath mitigation or equalization capabilities. Omnidirectional 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.

1.2.2 Directional Antennas

A single antenna can also be constructed to have certain fixed preferential transmission and reception directions. As an alternative to the brute force method of adding new transmitter sites, many conventional antenna towers today split, or sectorized cells. A 360° area is often split into three 120° subdivisions, each of which is covered by a slightly less broadcast method of transmission as shown in fig 1.2. All else being equal, sector antennas provide increased gain over a restricted range of azimuths as compared to an omnidirectional 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 sectorized antennas multiply the use of channels, they do not overcome the major disadvantages of standard omnidirectional antenna broadcast such as cochannel interference,

Figure 1.2Directional Antenna and Coverage Pattern [1]

1.3 Antenna Systems

An antenna can be made more intelligent by two ways. 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.

1.3.1Sectorized Systems

Sectorized antenna systems 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 omnidirectional 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. The covered area of a sectorized antenna is presented in fig 1.3.

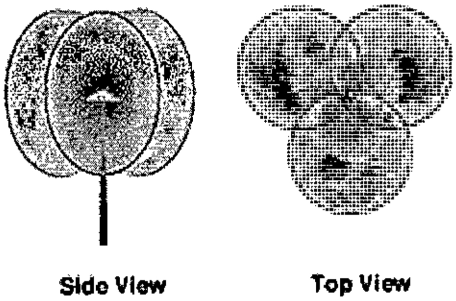

Figure 1.3 Sectorized Antenna and Coverage Patterns [1]

1.3.2Diversity Systems

In the next step toward smart antennas, the diversity system 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 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).

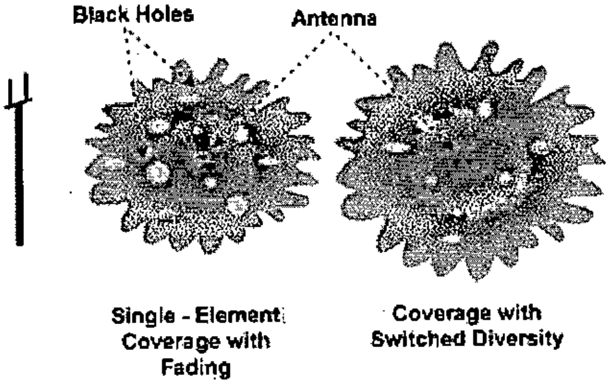

Figure 1.4 Switched Diversity Coverage with Fading and Switched Diversity [1]

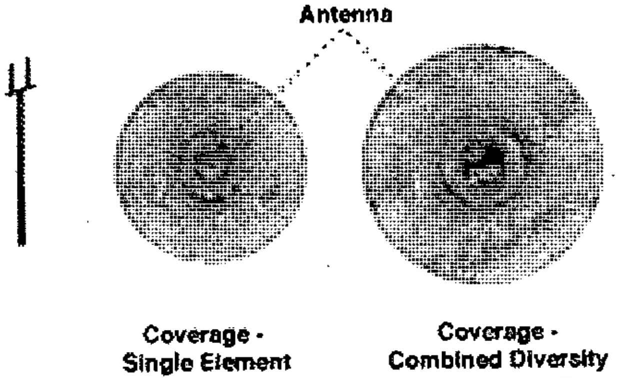

Figure 1.5 Combined Diversity Effective Coverage Pattern with Single Element and Combined Diversity [1]

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. From the fig 1.4 and 1.5 it is clears that coverage pattern of a combined diversity system is better than the switched diversity system.

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.

1.4. Smart Antenna System

In truth, antennas are not smart—antenna systems are smart. 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.

1.4.1 Types of Smart Antenna Systems

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 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

1.4.1.1 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 1.6 presents the formation of beams of a switched beam antenna array.

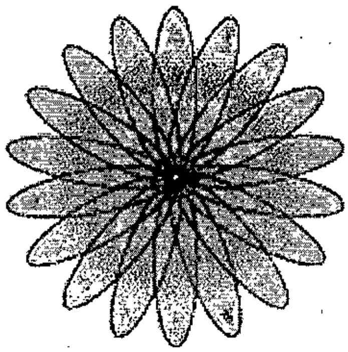

Figure 1.6 Switched Beam System Coverage Patterns (Sectors) [1]

1.4.1.2 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. Fig. 1.7 comprises the intelligence of an adaptive array coverage pattern.

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.

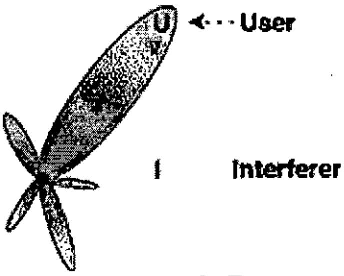

Figure 1.7 Adaptive Array Coverage: A Representative Depiction of a Main Lobe Extending Toward a User with a Null Directed Toward a Cochannel Interferer [1]

1.4.1.3 Space division multiple Access (SDMA)

As shown in Figure 1.8 from by using smart antenna arrays, we can now use space division multiple access (SDMA). In this case, users may use the same frequency, time, or code allocations over the air interface and only be separated spatially. This enables SDMA to be a complementary scheme to FDMA, TDMA, and CDMA, and SDMA thus provides increased capacity within congested areas.

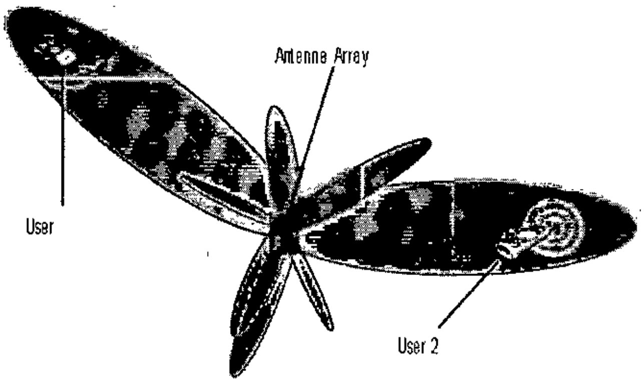

Fig1.8 SDMA allows two user to access the same base station on the same frequency [2]

1.4.2 Physical structure

Omnidirectional 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 laptop computers.

1.4.3 Key features

A simple antenna works for a simple RF environment. Smart antenna solutions are required as the number of users, interference, and propagation complexity grow. Their smarts reside in their digital signal-processing facilities.

Like most modern advances in electronics today, the digital format for manipulating the RF data offers numerous advantages in terms of accuracy and flexibility of operation. Speech starts and ends as analog information. Along the way, however, smart antenna systems capture, convert, and modulate analog signals for transmission as digital signals and reconvert them to analog information on the other end.

In adaptive antenna systems, this fundamental signal-processing capability is augmented by advanced techniques (algorithms) that are applied to control operation in the presence of complicated combinations of operating conditions.

The benefit of maintaining a more focused and efficient use of the system's power and spectrum allocation can be significant.

1.4.4 Intelligence of Smart Antenna

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 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.

1.4.5 The Goals of a 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 1.1*.

Table 1.1: Features and Benefits of Smart Antenna Systems

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 cochannel 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	**multipath rejection**—can reduce the effective delay spread of the channel, allowing higher bit

the array is used to minimize fading and other undesirable effects of multipath propagation.	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 the user)	**Reduced expense**—Lower amplifier costs, power consumption, and higher reliability will result.

1.5. Signal Propagation: Multipath and Cochannel Interference

1.5.1 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 omnidirectional 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 cochannel 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.

1.5.2 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.

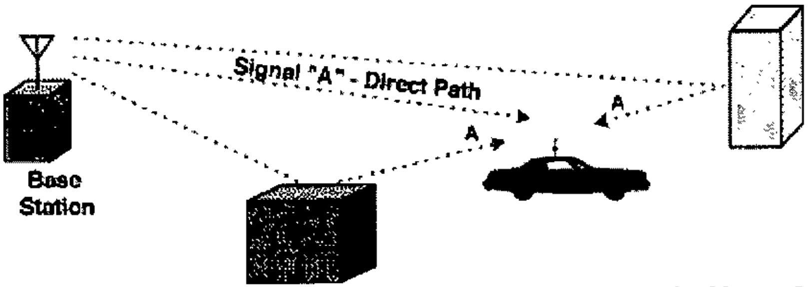

Figure 1.9 The Effect of Multipath on a Mobile User [1]

1.5.3 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. Figure 1.10 illustrates two out-of-phase signals as seen by the receiver.

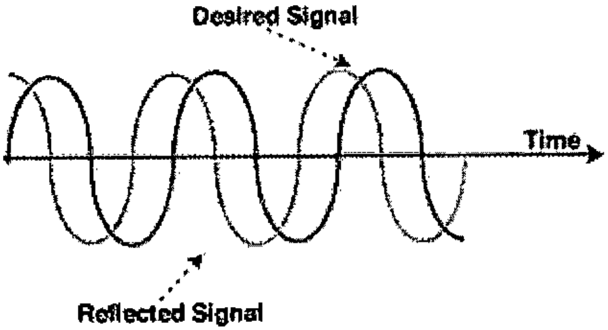

Figure 1.10 Two Out-of-Phase Multipath Signals [1]

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.

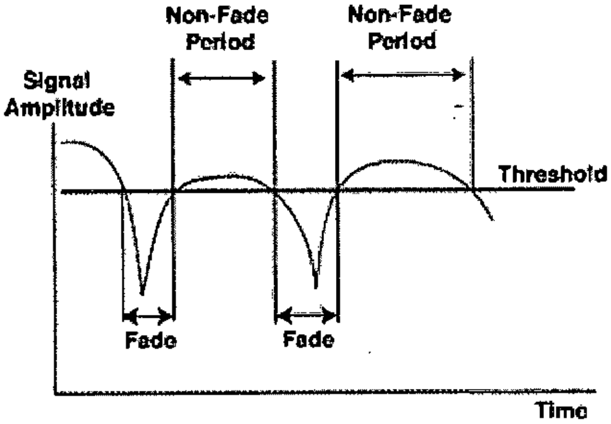

Figure 1.11 A Representation of the Rayleigh Fade Effect on a User Signal [1]

- **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.

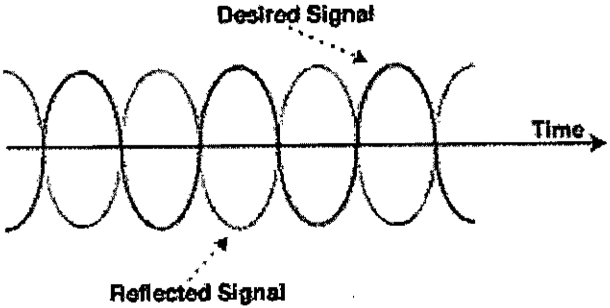

Figure 1.12 Illustration of Phase Cancellation [1]

- **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 intersymbol 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.

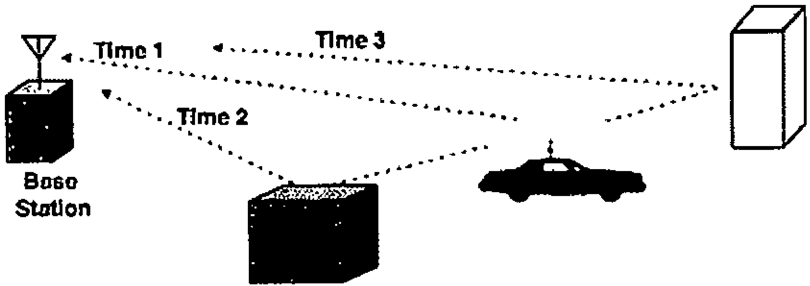

Figure 1.13 Multipath: The Cause of Delay Spread [1]

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.

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

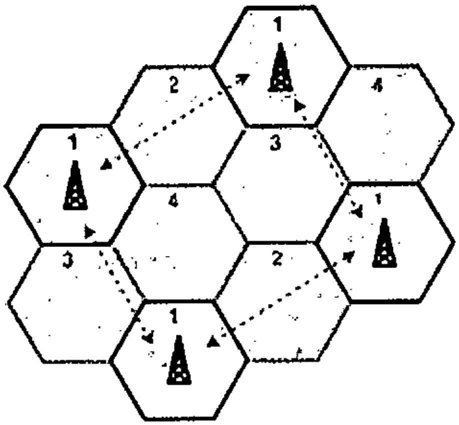

Figure 1.14 Illustration of Cochannel Interference in a Typical Cellular Grid [1]

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—cochannel interference. Management of cochannel interference is the number-one limiting factor in maximizing the capacity of a wireless system. To combat the effects of cochannel interference, smart antenna systems not only focus directionally on intended users, but in many cases direct nulls or intentional noninterference toward known, undesired users

1.6. The Architecture of Smart Antenna Systems

1.6.1 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.

1.6.2 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 omnidirectional antennas, leaving it to the base station to selectively 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.

1.6.3 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 (e.g., PHS and DECT) 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.

1.6.4 Switched Beam Systems

In terms of radiation patterns, switched beam 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 baseband 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.

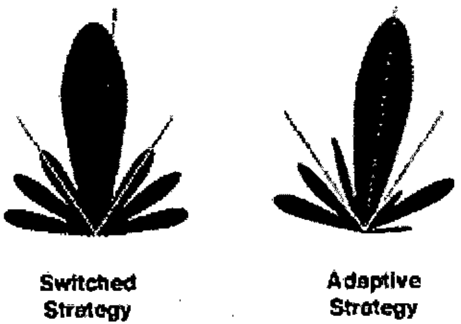

Figure 1.15 Beam forming Lobes and Nulls that Switched Beam (Red) and Adaptive Array (Blue) Systems Might Choose for Identical User Signals (Green Line) and Cochannel Interferers (Yellow Lines) [1]

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.

1.6.5 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 it's 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 1.16 illustrates the relative coverage area for conventional sectorized, switched beam, and adaptive antenna systems. 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.

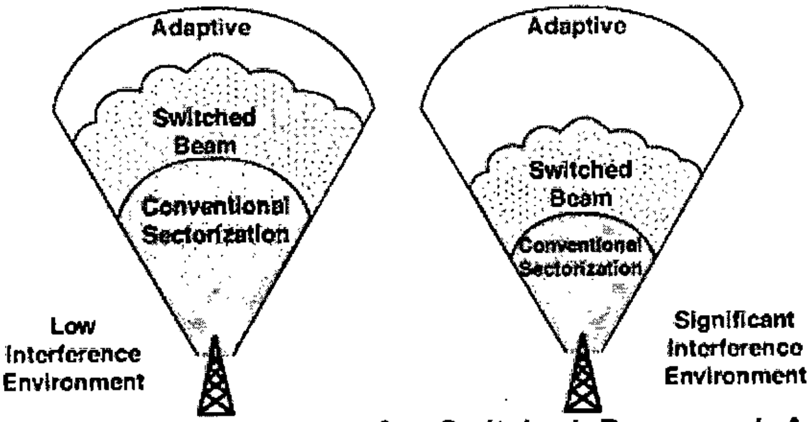

Figure 1.16 Coverage Patterns for Switched Beam and Adaptive Array Antennas

1.6.6 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 cochannel 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.

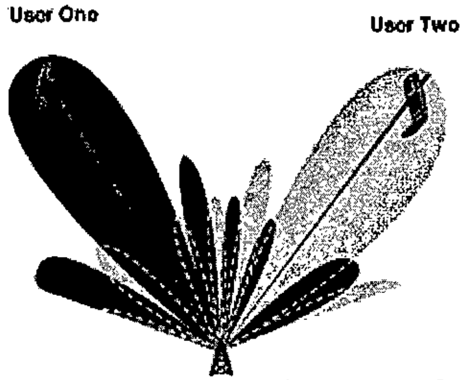

Figure 1.17 Fully Adaptive Spatial Processing, Supporting Two Users on the Same Conventional Channel Simultaneously in the Same Cell [1]

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.

1. 7 Technology used in Smart Antenna

Smart antenna technology can significantly improve wireless system performance and economics for a range of potential users. It enables operators of PCS, cellular, and wireless local loop (WLL) networks to realize significant increases in signal quality, capacity, and coverage.

Operators often require different combinations of these advantages at different times. As a result, those systems offering the most flexibility in terms of configuration and upgradeability are often the most cost-effective long-term solutions.

1.7.1 Applicable Standards

Smart antenna systems are applicable, with some modifications, to all major wireless protocols and standards, including those in *Table 1.2.*

Table1. 2: Applicable Standards

access methods	**analog**—frequency division multiple access (FDMA) (e.g., AMPS, TACS, NMT) **digital**—time division multiple access (TDMA) (e.g., GSM, IS–136); code division multiple access (CDMA) (e.g., IS–95)
duplex methods	frequency division duplex(FDD); time division duplex (TDD)

1.7.2 Transparency to the Network

The flexibility of adaptive smart antenna technology allows for the creation of new value-added products and services that give operators a significant competitive advantage. Adaptive smart antennas are not restricted to any particular modulation format or air-interface protocol. They are compatible with all current air-interface modulation schemes.

1.7.3 Added Advantages of Spatial Processing

A wide range of wireless communication systems may benefit from spatial processing, including high-mobility cellular systems, low-mobility short-range systems, wireless local loop applications, satellite communications, and wireless LAN. By employing an array of antennas, it is possible to multiplex channels in the spatial dimension just as in the frequency and time dimensions. To increase system capacity, spatially selective transmission as well as spatially selective reception must be achieved.

Improved algorithms and low-cost processors make sophisticated spatial processing practical alternatives for an increasing number of wireless system manufacturers and operators. Many agree that the unique benefits of spatial processing will ultimately affect all aspects of wireless system design.

Chapter -2

DOA Estimation Methods

2.1 Block diagram of Smart antenna array

The term smart antenna incorporates all situations in which a system is using an antenna array and the antenna pattern is dynamically adjusted by the system as required. Thus a system employing smart antennas processes signals induced on a sensor array. A block diagram of such a system is shown in fig 2.1

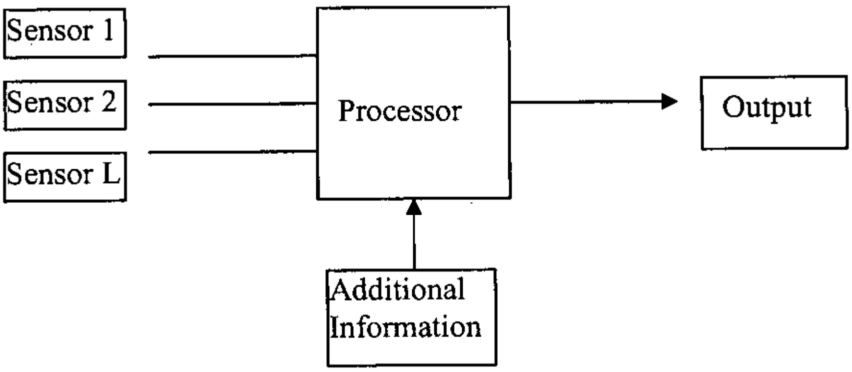

Fig 2.1 Block diagram of a smart antenna array

Additional information includes weight information, desired signal direction etc.

Consider the antenna array system consisting of L antenna elements shown in fig 2.2 where signal from each element are multiplied by a complex weight and summed from the array output which is expressed in a vector by using a co-ordinate system as shown in fig.2.3

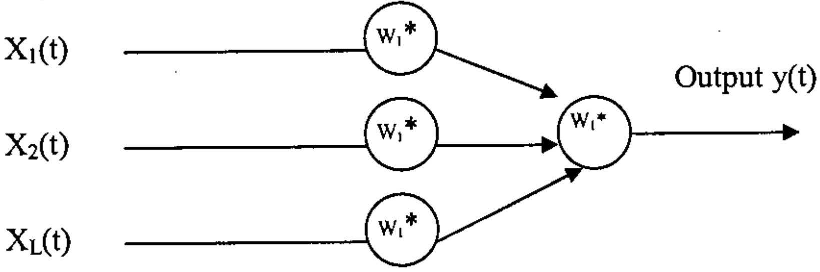

Fig 2.2 Antenna array system

The expression for the array output is given by

$$Y(t) = \sum W_L \cdot X_L(t) \tag{2.1.1}$$

W is weighting vector.

2.2 Linear Array and Beam Formation

Field pattern of a linear array consisting more than two elements can be controlled to achieve better directivity. Consider an n-element uniform linear array where all elements are spaced equally along a straight array as shown in Fig. 2.4 and the magnitude of the current in each element is the same and phase shift is progressive. Thus if the current in the element (k) is

$$I_k=I_0\, e^{jk\alpha} \qquad (2.2.1)$$

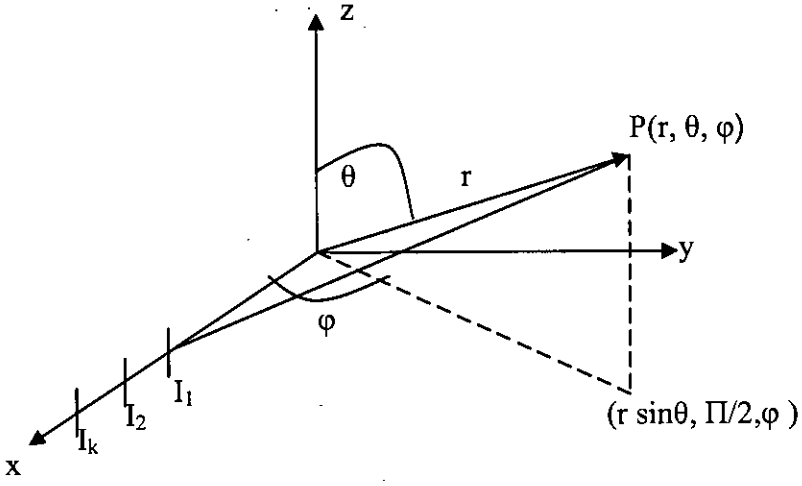

Fig 2.4 An n- element linear array

Then the current in the element (k+1) is

$$I_{k+1}= I_0 e^{j(k+1)\alpha} \qquad (2.2.2)$$

We have assumed that element (0) carries the reference current and has a maximum value I0. In the preceding equation, α is the progressive phase shift from one element to next.

Electric field density produced by a single element can be expressed from [3]

$$E_\theta= 1/r\ F(\theta,\ \varphi)e^{-j\beta r}\ e^{j(n-1)\psi/2}[\sin(n\psi/2)/\sin(\psi/2)] \qquad (2.2.3)$$

Where $\psi=\beta d\sin\theta\cos\varphi + \alpha$ (2.2.4)

Thus the normalize array pattern is

$$F(\psi)= [\sin(n\psi/2)/\sin(\psi/2)] \qquad (2.2.5)$$

In the xy plane (the plane perpendicular to the axis of the array) $\theta=90^0$ and $F(\theta, \varphi)=1$ for the both Hertzian dipole and the half-wave antennas. Thus, the field pattern depends only upon $F(\psi)$ at $\theta=90^0$.

The maximum value of $F(\psi)$ in (2.2.5) is n and it occurs at $\psi=0^0$. This is referred to as the principle maximum of the array. For a fixed point of observation $P(r, \pi/2, \varphi)$, φ is fixed. The progressive phase shift when $\psi=0^0$ is

$$\alpha= \beta d\cos\varphi \qquad (2.2.6)$$

By setting (2.2.5) to zero we obtain values for ψ for which the field intensity is zero each of these points is called null of the pattern. The null point occur when

$$\Psi=2p\,\pi/n \qquad p=1,2,3...... \qquad (2.2.7)$$

Between any two consecutive null points, the field pattens exhibit a secondary maximum point. We can obtain these points by setting $\sin(n\psi/2)=1$ That is

$$\Psi=(2q+1)\,\pi/n \qquad q=1,2,3.... \qquad (2.2.8)$$

The secondary maximum from (2.2.8) occurs when $\Psi=(2q+1)\,\pi/n$

And the amplitude of the first secondary maximum lobe from (2.2.5) is $1/\sin(1.5\,\pi/n)$, when n is very large. It is clear that the number of secondary maxima or side lobes q can be infinite but the number of elements is always finite. So the number of beams may not equal to the number of elements in the array.

2.3 Field pattern of a 20- Element Linear Array:

Consider $\alpha=0^0$ $\beta d=\pi/4$ and $n=20$

a) Field pattern in the xy plane

$\Theta=90^0$ $F(\theta,\varphi)=1$

$\Psi= \pi/4 \cos(\varphi)$

$F(\Psi)=\sin(10\Psi)/\sin(0.5\Psi)$

Setting $\Psi=0$, we find that the principle maxima are along $\varphi=90^0$ and $\varphi= 270^0$. The field pattern in the xy plane is shown in fig.2.5 Note that when the currents are in phase and the antennas are arranged along the x-axis, the principle lobe is in the y- direction ($\varphi=90^0$ and $\varphi= 270^0$)

When the field pattern is maximum in a direction perpendicular to the array, it is called a broadside array. To obtain a null in the x direction, the spacing between the elements must be $\lambda/2$. By setting $d= \lambda/2$ and $\alpha=\pi$, we can obtain the field pattern along the x-axis. An array that directs power along its direction is referred to as an end-fire-array.

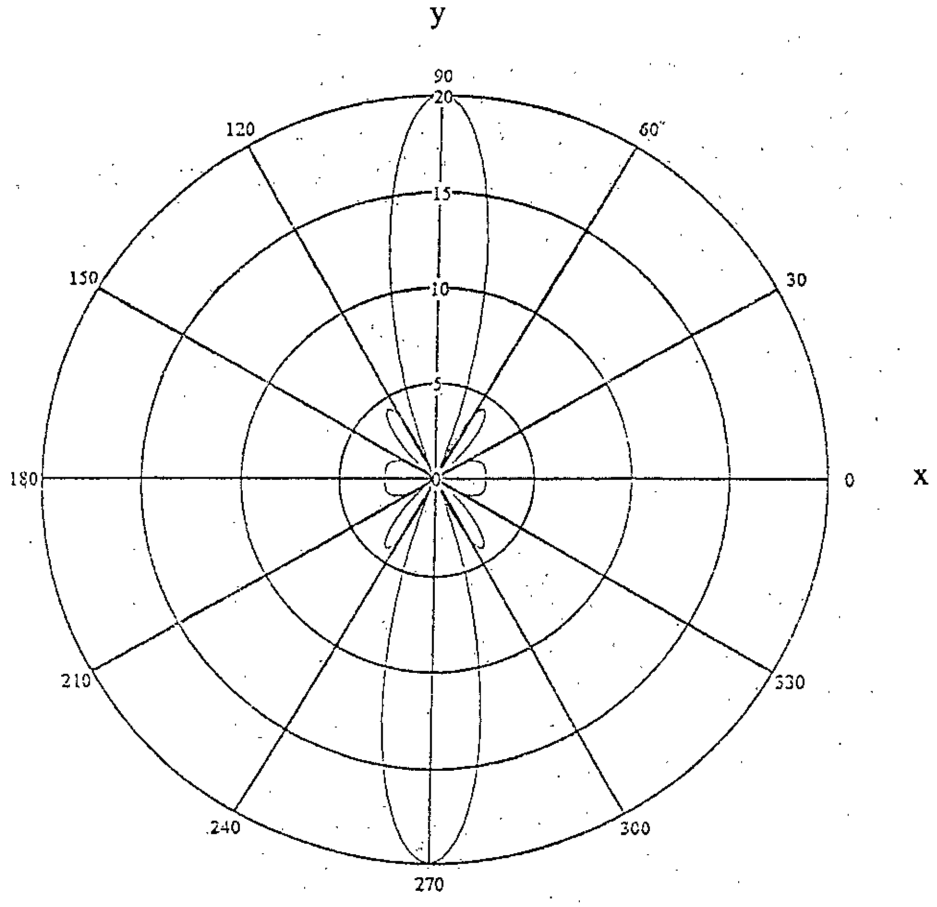

Fig2.5 Field pattern of a 20-element uniform array when d= λ/8, θ=90⁰ and α=0⁰

a) To obtain a Field pattern in the xz plane

$\varphi=0^0$ $F(\theta, \varphi) = \mathrm{Sin}(\theta)$

$\Psi = \pi/4\ \mathrm{Sin}(\theta)$

$F(\Psi)=\mathrm{Sin}(10\Psi)/\mathrm{Sin}(0.5\Psi)$

The field pattern in the xz plane is shown in fig.2.6

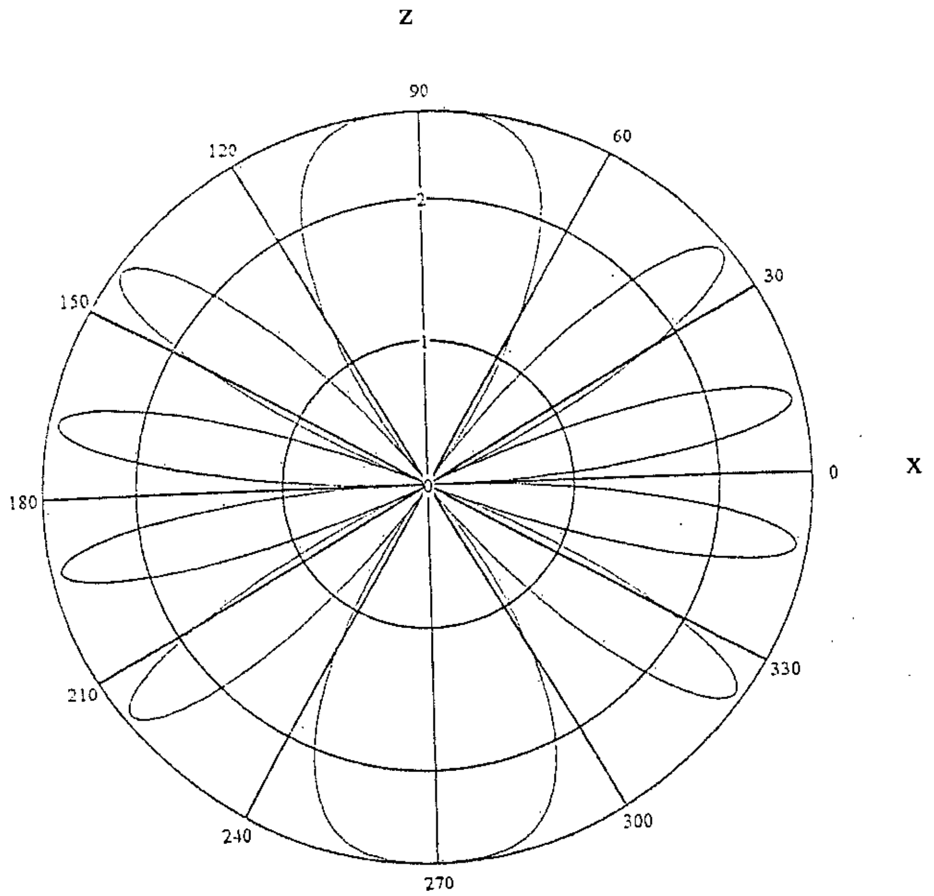

Fig 2.6 Field pattern of a 20-element uniform linear array in the xz plane when d= λ/8, $\varphi=0^0$ and $\alpha=0^0$

2.4 DOA estimation Methods:

The problem of localization of sources radiating energy by observing their signal received at spatially separated sensors is of considerable importance, occurring in many fields, including radar, sonar, Mobile communication. In this chapter an estimation of the direction of arrival (DOA) of narrowband sources of the same central frequency located in the far field of an array of sensors is considered and various DOA estimation methods are described, compared and sensitivity of various perturbations is analyzed.

The chapter also contains discussion of various pre-processing and source estimation methods. Source direction is parameterized by the variable Θ. The DOA estimation method considered includes spectral estimation, minimum variance, and distortion less response estimator, linear prediction, Maximum entropy, and Maximum likelihood. Various Eigen structure methods are also described, including many versions of MUSIC algorithm, minimum norm method, CLOSEST method, ESPRIT method and weighted subspace fitting method.

2.5 Spectral Estimation Method

These methods estimate DOA by computing the special spectrum P (θ) that is the mean power received by an array as a function of O and then determining the local maximum of this computed special spectrum. Most of the techniques have their roots in time series analysis.

2.5.1 Bartlett Method

One of the earliest methods of spectral analysis is the Bartlett method in which a rectangular window of uniform weighted is applied to the time series data to be analyzed. For bearing estimation problems using an array, this is equivalent to applying equal weighting to each element. Thus by steering the array in θ direction this method estimates the mean power P (θ), an expression for which is given by using [4]

$$P_B(\theta) = S_\theta^H R S_\theta / L^2 \qquad (2.3.1)$$

Where S_θ denotes the steering vector associated with the direction θ, L denotes the number of elements in the array and R is the correlation matrix.

A set of steering vector so associated with various direction θ is often referred to as the array manifold in DOA estimation literature. In practice it may be measured at the time of array calibration. From the array manifold and the estimate of the array co-relation matrix $P_B(\theta)$ is computed using (2.3.1). Peaks in $P_B(\theta)$ are then taken as the direction of the radiating sources.

The process is similar to that of mechanically steering the array in this direction and measuring the output power. Due to resulting side-lobes output power is not only contributed from the direction in which the array is steered but from the direction where the side lobes are pointing. The processor is also known as the conventional beam former and the resolving power of the processor depends on the aperture of the array or the beam width of the main-lobe.

2.5.2 Minimum Variance distortion less response estimator:

The Minimum Variance distortion less response estimator (MVDR) is the maximum likelihood method (MLM) of spectrum estimation which finds the maximum likelihood (ML) estimate of the power arriving from a point source in direction o assuming that all the sources are interference. In the beam forming literature it is known as the MVDR beam former as well as the optimal beam former, since in the absence of errors it maximize the output SNR and passes the look direction signal undistorted. For DOA estimation problems, MLM is used to find ML estimate of the direction rather than the power following this convention the current estimator is referred to as a MVDR estimator.

This methods uses the array weights obtained by minimizing the mean output power subject to a unity constrain in the look direction. The expression of the power spectrum $P_{MV}(\theta)$ is found in [5]

$$P_{MV}(\theta) = 1/ S_\theta^H R^{-1} S_\theta \qquad (2.3.2)$$

This method has better resolution properties than the Bartlett method but does not have the best resolution properties of all methods.

2.6 Linear prediction method:

The linear prediction (LP) method estimates the output of one sensor using linear combination of the remaining sensor outputs and minimizes the mean square prediction error that is the error between the estimate and the actual output. Thus it obtains the array weight by minimizing the mean output power of the subject to the constrain that weight on the selected sensor is unity. Expression for the array weight and the power spectrum $P_{LP}(\theta)$ respectively are presented from [4]

$$P_{LP}(\theta) = u_1^H R^{-1} u_1 / |u_1^H R^{-1} S_\theta|^2 \qquad (2.4.1)$$

Where u1 is column vector such that one of its element is unity and remaining elements are zero.

The position of 1 in the column vector corresponds to the position of the selected element in the array for predicting its output. There is no criterion for proper choice of this element. However choice of this element affects the resolution capability and bias in the estimate. These effects are dependent on SNR and separation of the directional sources. LP methods perform well in moderately low SNR environments and are good compromises in situation where sources are of approximately equal strength and are nearly coherent.

2.7 Maximum Entropy Method

The maximum entropy (ME) method finds a power spectrum such that its Fourier transform equals the measured correlation subjected to the constraint that its entropy is maximized. The entropy of a Gaussian band-limited time series with power spectrum S (f) is defined as used in [6]

$$H(s) = \int^{FN} \ln S(f)\, df \tag{2.5.1}$$

Where F_N is the Nyquist frequency.

For estimating DOA from the measurement using an array of sensors, the ME method finds a continuous function $P_{ME}(\theta) > 0$ such that it maximize the entropy function.

2.8 Maximum Likelihood Method:

The MLM estimates the DOAs from a given set of array samples by maximizing the log-likelihood function. The likelihood function is the joint probability density function of the sampled data given the DOAs and viewed as a function of the desired variables, which are the DOAs in this case. The method searches for those directions that maximize the log of this function. The ML criterion signifies that planes waves from these directions are most likely to cause the given samples to occur.

Maximization of the log-likelihood is a nonlinear optimization problem, and in the absence of a closed-form solution requires iterative schemes. There are many such schemes available in the literature. The well-known gradient decent algorithms using the estimated gradient of the function at each iteration as well as the standard Newton-Raphson method method are well suited for the job. Other schemes such as the alternating projection method and the expectation maximization algorithm have been proposed for solving this problem in general as well as for specialized cases such as unknown polarization, unknown noise environments and contaminated Gaussian noise. A fast algorithm based on Newton's method developed for estimating frequencies of sinusoids may be modified to suit DOA estimation based on ML criteria.

The MLM provides superior performance compared to other methods particularly when SNR is small, the number of samples is small, or the sources are correlated and thus is of practical interest. For a single sources the estimates obtained by this method are a asymptotically unbiased, that is the expected values of the estimates approach their true values in the limit as the number of samples used in the estimate increase .In that sense it may be used as a standard to compare the performance of other methods. The method normally assumes that the number of sources, M, is known.

When a large number of samples are available, other computationally more efficient schemes may be used with performance almost equal to this method. Analysis of the method to estimate the direction of sources when the array and the sources are in relative motion to each other indicates its potential for mobile communication.

2.9 Eigen structure Methods:

These methods rely on the following properties of the array correlation matrix: 1) The space spanned by its eigenvectors may be partitioned in two subspaces, namely the signal subspace and the noise subspace and 2) the steering vectors corresponding to the directional sources are orthogonal to the noise subspace. As the noise subspace is orthogonal to the subspace, these steering vectors are contained in the signal subspace. It should be noted that the noise subspace is spanned by the eigenvectors associated with the smaller Eigen vales of the correlation matrix, and the signal is spanned by eigenvectors associated with its larger Eigen values.

In principle the Eigen structure-based methods search for direction such that the steering vectors associated with this direction are orthogonal to the noise subspace and are contained in the signal subspace. In practice the search may be divided into two parts. First find a weight vector w that is contained in the noise subspace or is orthogonal to the signal subspace, and then search for direction such that the steering vectors associated with this direction are orthogonal to this vector. The sources directions correspond to the local minima of the function $|W^H S_\theta|$, where S_θ denotes a steering vector from [4].

When these steering vectors are not guaranteed to be in the signal subspace there may be more minima than number of sources. This distinction between the actual sources direction and spurious minima in $|W^H S_\theta|$ is made by measuring the power in this direction.

Many methods have been proposed that utilize the Eigen structure of the array co-relation matrix. These methods differ in the way that available array signals have been utilized, required array geometry. Applicable signal model and so on. Some of this method does not require explicit computation of the Eigen values and eigenvectors of the array co-relation matrix, whereas in others it is essential. Effective computation of these quantities may be done by methods similar to those descried. When the array co-relation matrix is not available, a suitable estimate of the matrix is made from available samples.

One of the earliest DOA estimation methods based on the Eigen structure of covariance matrix was presented in [3] and has better resolution than the minimum variance, ME and LP methods. A critical comparison of this method with two other schemes applicable for a co-related noise field has been presented to show that the Pisarenko's method is an economized version of these schemes restricted to equispace linear arrays.

Eigen structure methods may also be used for finding DOAs when the background noise is not white but has a known covariance or when the sources are in the rear field and/or the sensor have unknown gain pattern. For the later case the signal induced on all elements of the array are not of the equal intensity, as is the case when the array is in the far field of the directional sources. The effect of spatial coherence on resolution capability of these methods is discussed.

2.10 MUSIC Algorithm

The multiple signal classification (MUSIC) method is a relatively simple and efficient Eigen structure variant of DOA estimation methods. It is perhaps the most studied method in its class and has many variations. Some of these are discussed in this section

2.10.1Spectral MUSIC

In its standard form also known as spectral MUSIC, the method estimates the noise subspace form available samples. This can be done either by Eigen values decomposition of the estimated array correlation matrix or singular value decomposition of the data matrix with its N columns being the N array signal vector samples, also known as snapshot.

Once the noise subspace has been estimated, a search for M direction is made by looking for steering vectors that are as orthogonal to the noise subspace as possible. Thos is normally accomplished by searching for peaks in the MUSIC spectrum given in [7]

$$P_{MU}(\theta) = 1/|S_\theta^H U_N|^2 \qquad (2.8.1)$$

Where Un denotes an L by L-M dimensional matrix, with L-M column being the eigenvectors corresponding to the L-M smallest Eigen values of the array correlation matrix and SO denoting the steering vector that correspond to direction Q.

It should be noted that instead of using the noise subspace and searching of directions with steering vector orthogonal to the subspace, one could also use the signal subspace and search for directions with steering vectors contained in this space. This amount to searching for peaks is stated in [7]

$$P_{MU}(\theta) = |U_S^H S_\theta|^2 \qquad (2.8.2)$$

Where Us denotes an LXM dimensional matrix with its M columns being the eigenvectors corresponding to the M largest Eigen values of the array correlation matrix.

It is advantageous to use the one with smaller dimensions. For the case of a single source, the DOA estimate made by the MUSIC method asymptotically approaches the Cramer-Rao lower bound, that is, where the number of snapshots increase infinitely, the best possible estimate is made. For multiple sources, the same holds for large SNR cases, That is, when the SNR approaches infinity. The Cramers-Rao lower bound (CRLB) gives the theoretical lowest values of the covariance for an unbiased estimator.

In an application of the MUSIC algorithm to cellular mobile communication was investigated to locate land mobiles and it is shown that when multipath arrivals are grouped in cluster the algorithm is able to locate the mean of each cluster arriving at a mobile. This information then may be used to locate line of sight. Its use for mobile satellite communications has been suggested.

2.10.2Root-MUSIC

For a uniformly spaced linear array (ULA), the MUSIC spectra can be expressed such that the search for DOA can be made by finding the roots of a polynomial. In this case, the method is known as root-MUSIC. Thus, root-MUSIC is applicable when a ULA is used and solves the polynomial rooting problem in contrast to spectral Music's identification and localization of spectral peaks. Root-MUSIC has better performance than spectral MUSIC.

2.10.3 Constrained MUSIC:

This method incorporates the known source to improve estimates of the unknown source direction. The situation arises when some of the source directions are already known. The method removes signal components induced by these known sources from the data matrix and then uses the modified data matrix for DOA estimation. Estimation is achieved by projecting the data matrix into a space orthogonal complement to a space spanned by the steering vectors associated with known source directions. A matrix operation, the process reduce the signal subspace dimension by a number equal to known sources and improves estimate quality, particularly when known sources are strong or co-related with unknown sources.

2.10.4 Beam Space MUSIC:

The MUSIC algorithm discussed so far process the snapshots received from sensors elements without any preprocessing , such as forming beams and thus may be thought of as element space algorithm, which contrasts with the beam space MUSIC algorithm in which the array data are passed through a bemforming processor before applying MUSIC or any other DOA estimation algorithms The beam forming processor output may be thought of as asset of beams, thus the processing using this data is normally referred to as beam space processing. A number of DOA estimation schemes are discussed in [4] where data are obtained by forming multiple using an array.

The DOA estimation in beam space has a number of advantages such as reduced computation, improved resolution, reduced sensitivity of system errors, reduced resolution threshold, and reduced bias in the estimate and so on. These advantages arise from the fact that a beam former is used to form a number of beams that are less than the number of elements in the array. Consequently less data to process DOA estimation is necessary.

This process may be understood in terms of array degrees of freedom. Element space methods have degrees of freedom equal to the number of elements in the array, whereas the degrees of freedom of beam space methods are equal to the number of beams formed by the beam forming filter. Thus, the process reduces the array's degrees of freedom. Normally only M+1 degrees of freedom are needed to resolve M sources.

The root-MUSIC algorithm discussed for the element space case may be applied to this case, giving rise to beam space root-MUSIC. Computational savings for this method are the same as for beam space methods compared to element space methods in general.

2.11 Minimum Norm Method

Minimum norm method is applicable for ULA and finds the DOA estimates by searching the peak location in the spectrum as in the following expression found in [8]

$$P_{MN}(\theta) = 1/|W^H S_\theta|^2 \qquad (2.9.1)$$

Where w denotes the array weight such that it is of the minimum norm, has first element equal to unity and is contained in the noise subspace. [8] Gives the solution to the above problem leads to the following expression for the spectrum.

$$P_{MN}(\theta) = 1/|U_N S_\theta^H U_N^H e_1|^2 \qquad (2.9.2)$$

Where the vector e1 contains all zeroes except the first element, which is equal to unity.

Given that the method is applicable for ULA, the optimization problem to solve for the array weight may be transformed to a polynomial rooting problem, leading to a root-minimum-norm method similar to root MUSIC. A performance comparison indicated that the variance in the estimate obtained by the root MUSIC is smaller than or equal to that of root- minimum-norm method. Schemes to speed up the DOA estimation algorithm of the minimum-norm and to reduce computation are discussed in [5]

2.12CLOSEST Method:

The CLOSEST method is useful for locating sources in a selected sector. Contrary to beam space methods, which work by first forming beams in selected directions, CLOSEST operates in the element space and in that sense it is an alternative to beam space MUSIC. In a way it is a generalization of the minimum norm method. It searches for array weights in the noise subspace that are close to steering vectors corresponding to DOAs in the sector under consideration, and thus its name. Depending on the definition of closeness, it leads to various schemes. A method referred to as FINE (First principle vector) selects an array weight vector by minimizing the angle between the selected vector and the subspace spanned by the steering vectors corresponding to DOAs in the selected sector.

In short the method replaces the vector e1 used in the minimum norm method by able vector depending on the definition of closeness used. For details about the selection of these vectors and the relative merits of CLOSEST method are found in [9]

2.13 ESPRIT Method

Estimation of signal parameters via rotational invariance technique (ESPRIT) is computationally efficient and robust method of DOA estimation. It uses two identical arrays in the sense that array elements need to form matched pair with an identical displacement vector, that is, the second element of each pair ought to be displaced by the same distance and in the same direction relative to first element.

However, this does not mean that one has to have two separate arrays. The array geometry should be such that the elements could be selected to have this property. Foe example a ULA for four identical elements with inter element spacing d may be thought of as two arrays of three matched pairs, one with first three element and second with last three element such that the first and the second elements form pair, the second and the third element from another pair and so on. The two arrays are displaced by the distance d. the way ESPRIT exploits this subarray structure for DOA estimation is found briefly described in [10].

2.14 Weighted subspace fitting method

The Weighted subspace fitting method (WSF) is a unified approach to schemes such as MLM, MUSIC and ESPIRIT. It requires that the number of directional sources be known. The method finds the DOA such that the weighted version of a matrix whose columns are the steering vectors associated with this directions is close a data-dependent matrix. The data-dependent matrix could be a Hermitian square root of the array correlation matrix or a matrix whose columns are the Eigen vectors associated with the large Eigen values of the array correlation matrix. The framework proposed in the method can be used for driving common numerical algorithm for various Eigen structure methods as well as for their performance studies. WSF application for mobile communication employing an array at the base station has been investigated in [11].

2.15 Review of Other Methods

So far discussed above a number of Eigen structure methods reported in the literature exploited specialized array structure or noise scenarios. Two methods using uniform circular arrays presented in extend beam space MUSIC or ESPRIT algorithm for two dimensional angle estimation, including an analysis of MUSIC to resolve two sources in the presence of gain, phase and location errors. Properties of an array have also been exploited in to find the azimuth and elevation of directional source. Two DOA estimation schemes in an unknown noise field using two separate arrays proposed in appear to offer superior performance compared to their conventional counterparts.

Use of minimum redundancy linear array offers several advantages are discussed. By using such arrays one may be able to resolve more than L sources using L elements, L (L-1)/2 being the upper limit. A minimum redundancy linear array has no uniform spacing such that the number of sensors pairs measuring the same special correlation lag is minimized for a given number of elements. In designing such an array having only one pair with spacing d, one pair with spacing 2d, and so on is preferred, such as a three element array with element position x1=0, x2=d, x3=3d. The minimum redundancy linear arrays are also referred to as augmented arrays.

The direction-finding methods applicable to unknown noise field are described. The MAP (maximum a posteriori) method presented is based on Bayesian analysis and estimated results are not asymptotically consistent, that is the result may be biased. The method referred to as concurrent mulling and location (CANAL) may be implemented using hardware, thus eliminating the need for sampling, data storage and so on. A DOA estimation method in the presence of correlated arrivals using an array of unrestricted geometry is discussed in [12].

Chapter -3

Processing Techniques, Performance & Sensitivity

3.1 Smart Antenna Processing

A fully adaptive antenna array implementation requires a considerable increase in processing requirements. Previously, we had a single stream of data coming from a single antenna. Now, we have multiple data streams to process. As shown in Figure 3.1 from [2] the data flow diagram for a beam- forming application is not a single input data stream. We now have N data streams that must be processed from the N antenna elements. Antenna elements in adaptive arrays is to pass the data stream from each antenna through an adaptive finite impulse response (FIR) filter. Note that in narrowband applications, the adaptive FIR filters simplify to
a single weight vector. The processing requirements increase, however, with each beam processed
If we consider a simple example where we have four antennas and a narrowband system, such that the adaptive filters result in a single multiplication, we can see that the processing requirements approach one- half billion multiple accumulates MACs per second, for a sample rate of 105 mega samples per second. This sample rate is for a single beam and does not include the processing requirements for the adaptive update algorithm. This amount of processing does not seem unreasonable for performance in a DSP processor. However, if we want to support multiple beams and achieve finer beams by increasing the number of antennas, we could quickly exhaust the processing capability of a standard processor architecture as we reach processing requirements of several billion MACs per second.
By using FPGAs, we have powerful DSP devices for handling these high-performance requirements at sampled data rates. Furthermore, we can take advantage of the FPGA flexibility for directly handling acquisition control and other DSP functions, such as digital down-conversion, demodulation, and matched filtering

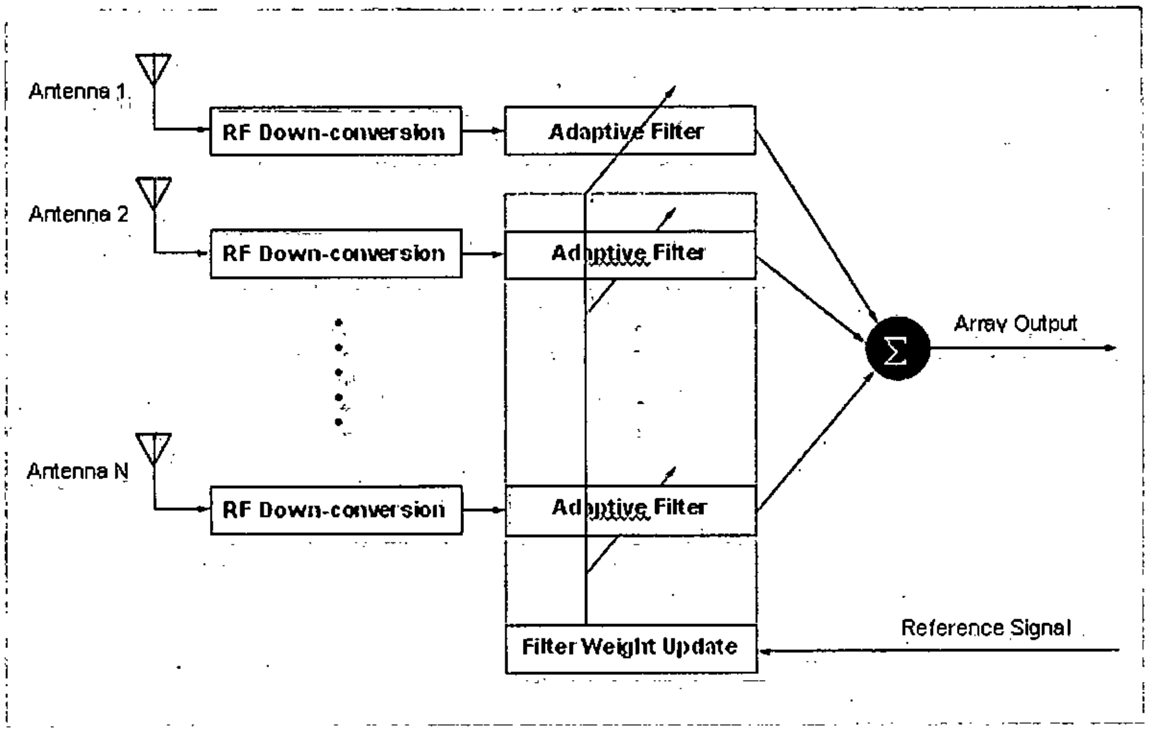

Fig: 3.1 Data flow for beam forming [2]

3.2 Processing Techniques

Several techniques are used to process data before using direction finding methods for DOA estimation, particularly in situation where directional sources are correlated or coherent. Correlation of directional sources may exist due to multipath propagation and tends to reduce the rank of the array correlation matrix. The correlation matrix may be tested for source coherency by applying the rank profile. Most pre-processing technique either tries to restore the rank deficiency in the correlation matrix or modify it to be useful for the DOA estimation methods.

Special smoothing method for beam forming decor relates the correlated arrival by subdividing the array into a number of smaller overlapping subarrays and then averaging the array correlation matrix obtained from each subarray. The number of sub-arrays obtained from an array depends on the number of element used in each sub-array. For example, using k element on each sub-array L-(k-1) subarrays can be formed from an array of L elements by forming the first sub-array using element 1 to k, the second subarray using elements 2 to k+1 and so on. The number and size of sub-arrays are determined from the number of directional sources under consideration. For M sources, a sub-array size of M+1 and a subarray number greater than or equal to M are necessary.

Thus to estimate the directions of M sources, array size L=2M is required, which could be reduced to 3/2M by using forward-backward spatial smoothing method. This process use the average of the correlation matrix obtained from the forward sub-array scheme and the correlation matrix obtained from the backward subarray scheme.

The forward subarray scheme subdivides the array starting from one side of the array as discussed above, whereas the backward subarray schemed subdivides the array starting from the other side of the array. Thus in the forward subarray scheme , the first subarray is formed using element 1 to k whereas the backward subarray scheme the first subarray scheme is formed using elements L to L-(k-1) and so on.

A method described in [7] removes the effects of sensor noise to make spatial smoothing more effective in low SNR situation. This spatial filtering method is further refined in [10] to offer DOA estimates of coherent sources with reduced RMS errors.

A decorrelation analysis of spatial smoothing shows that there exists an upper bound on the number of subarrays and the maximum distance between the subarrays depends on the fractional bandwidth of the signals. A comprehensive analysis of the use of spatial smoothing as a preprocessing technique to weighted ESPRIT and MUSIC methods of DOA estimation presented in [9] shows how their performance could be improved by proper choice of their number of subarrays and weighting matrices. An ESPRIT application to estimate the sources directions and polarization shows improvement in its performance in the presence of coherent arrivals when it is combined with the spatial smoothing method.

Spatial smoothing methods using subarray arrangements reduce the effective aperture of the array as well as degrees of freedom, and thus more elements are needed to process correlated arrival than would otherwise be required. The schemes that do not reduce effective array size include those that restore the structure of the array correlation matrix for the linear array to an uncorrelated one.

Structured methods rely on the fact that for a linear equispaced array, the correlation matrix in the absence of arrivals has a Toeplitz structure, that is, the elements of the matrix along its diagonals are equal. Correlation between sources destroys this structure. The structure is restored by averaging the matrix obtained in the presence of correlated arrivals by simple averaging along the diagonals as detailed while in a weighted average is used. A DOA estimation method using the array correlation matrix structured by averaging along its diagonal discussed in [8] appears to offer computational advantages over similar methods

Others preprocessing techniques to decorrelate sources include random permutation, mechanical movement using a circular disk, construction of preprocessing matrix using approximate knowledge of DOA estimate, signal subspace transformation method and method based on aperture interpolations.

3.3 Estimating Source Number

Many high-resolution direction-finding methods require that the number of directional sources and their performance is dependent on perfect knowledge of these numbers. Selected methods for estimating the number of these sources are discussed in this section.

The most commonly referred method for detecting the number of sources was first introduced in [6] based on Akaikes information creation (AIC) and Rissanen's minimum description length (MDL) principle. This method was further analyzed and modified. A variation of the method that is applicable to co herent sources is as follows

1. Estimate the array correlation matrix from N independent and identically distributed samples.
2. Find the Eigen values $\lambda i=1,2.....L$ of the correlation matrix such that $\lambda 1 > \lambda 2 > ... > \lambda_L$
3. Estimate the number of sources M by solving

$$\text{Minimize } N(L-M)\log(f1(M)/f2(M))+f3(M,N)$$

With L denoting the number of elements in the array.

A modification of the method based on the MDL principle applicable to coherent sources is discussed which is further refined to improve performance. A parametric method that does not require knowledge of Eigen values of the array correlation matrix is discussed in [13]. It has better performance than some other methods discussed and is computationally more complex.

All methods that partition the Eigen values of the correlation rely on the fact that the M Eigen values corresponding to M directional sources are larger than the rest of the L-M Eigen values corresponding to the background noise, they also select the threshold differently. One of the earliest methods uses a hypothesis-testing procedure based on the confidence interval of noise Eigen values. Threshold assignment was subjective.

The Eigen threshold method uses a one-step prediction of threshold for differentiating the smallest eigenvaluesfrom the others. This methods performs better than AIC and MDL. It has a threshold at a lower SNR value than MDL and a lower error rate than AIC at high SNRs.

An alternate scheme for estimating the number of sources discussed in [lee94] uses the eigenvectors of the array correlation matrix; in contrast, other methods use the Eigen values of the array correlation matrix. This method referred to as the eigenvector detection technique, is applicable to a cluster of sources whose approximate direction are known.

In practice the number of sources of an array may be able to resolve not only depends on the number of elements in the array but also the array geometry, available number of snapshot, and spatial distribution of sources. For discussion of these and other issues related to array capabilities to uniquely resolve the number of sources.

3.4 Performance Comparison

Performance analysis of various direction finding-schemes has been carried out by many researches. The performance measures considered for analysis include bias, variance, resolution, CRLB and probability of resolution. This section, the performance of selected estimation schemes is discussed.

Most of the studies of MUSIC algorithm concentrate on its performance and performance comparisons with other methods when a finite number of samples are used for direction finding rather than their ensemble average.

A rigorous bias analysis of MUSIC shows in [2] that the MUSIC estimates are biased. For a linear array in the presence of a single source, the bias increase as the sources moves away from broadside. Interestingly the bias also increases as the number of elements increases without changing the aperture. An asymptotic analysis of MUSIC with forward-backward spatial smoothing in the presence of correlated arrivals shows that to estimate two angles of arrival of equal power under identical conditions, more snapshots are required for correlated sources than for uncorrelated sources.

Bias and the standard deviation (STD) are complicated functions of the array geometry, SNR, and number and directions of sources and vary inversely as the number snapshots. A poorer estimate generally results using a smaller number of snapshots and sources with lower SNR. The performance of conventional MUSIC is poor in the presence of correlated arrivals, and it fails to resolve coherent sources.

Although the bias and STD both play important roles in direction estimation, the effect of bias near the threshold region is critical. A comparison of MUSIC performance with those of the minimum-norm and FINE for finite-sample cases shows that in the low SNR range, the minimum norm estimates have the largest STD and MUSIC estimates have the largest bias. These results are dependent on sources SNR is increased. The overall performance of FINE is better than the other two in the absence of correlated arrivals.

The estimates obtained by MUSIC and ML methods are compared with the CRLB for large-sample cases. The CRLB gives the theoretically lowest values of the covariance of an unbiased estimator, it decrease with the number of samples, number of sensors in the array and source SNR. The study concluded that the MUSIC estimates are the large-sample realization of ML estimates in the presence of uncork-related arrivals. Furthermore it shows that the variance of the MUSIC estimate is greater than that of the ML estimates and the variance of the two methods approaches each other as the number of elements and snapshot increases. Thus using a large number of elements and samples excellent estimates is possible of direction of uncorrelated sources with large SNRs using the MUSIC method. It should be noted that MLM estimates are unbiased. An unbiased estimate is also referred as a consistent estimate.

An improvement in MUSIC DOA estimation is possible by beam space MUSIC. By properly selecting a beam forming matrix and then using the MUSIC scheme to estimate DOA one is able to reduce the threshold level of the required SNR to resolve the closely space sources. Although the variance of this estimate is not much different from the element space case, it has less bias. The resolution threshold of beam space MUSIC is lower than the conventional minimum- norm method. However for two closely space sources the beam space MUSIC and beam space minimum norm method provide identical performance when suitable beam forming matrix is selected.

When beam forming weights have conjugate symmetry the beam space MUSIC has decorrelation properties similar to backward-forward smoothing and thus is useful for estimation of correlated arrival source direction and offers performance advantages in terms of lower variance for the estimated angle.

The resolution property of MUSIC shows how it depends on SNR, number of snapshots, array geometry and separation angle of the two sources. The two closely spaced are said to be resolved when two peaks in the spectrum appear in the vicinity of the two sources direction. Analytical expression for resolution probability and its variation as function of various parameters are presented in [9] and could be used to predict the behavior of MUSIC estimates for a given scenario.

A performance comparison of MUSIC and other eigenvector method which uses noise eigenvectors divided by corresponding Eigen values for DOA estimation indicates that the former is more sensitive to the choice of assumed number of sources compared to actual number of sources.

Performance analysis of many versions of ESPRIT are considered and compared with other methods in different studies. Estimate obtained by subspace rotation methods that include the Toeplitz approximation method (TAM) and ESPRIT have greater variance than those obtained by MUSIC using large number of samples. Estimates by ESPRIT using a uniform circular array are asymptotically unbiased. LS-ESPRIT and TAM estimates are statically equivalent LS-ESPRIT and TLS_ESPRIT have the same MSE and their performance depends how subarrays are selected. The minimum norm method is equivalent to TLS-ESPRIT and root MUSIC outperform the ESPRIT. TAM is based on the state space model and finds DOA estimates for signal subspace. In sprit its approach is similar to ESPRIT. The WSF and ML are efficient for Gaussian signal as both attain CRLB asymptotically. A method is said to be efficient when it achieves CRLB.

A correlation between sources affects the capabilities of various DOA estimation algorithms differently. A study of the effect of the correlation between two sources on the accuracy of DOA finding scheme shows that the correlation phase is more significant than correlation magnitude. Most performance analysis discussed assumes that the background noise is white. When this is not than case the DOA scheme performs differently. In the presence of colored background noise MUSIC performance is better than that of ESPRIT and the minimum norm method over a wide range of SNR. The performance of minimum norm method is worse than MUSIC and ESPRIT.

3.5 Sensitivity analysis:

Sensitivity analysis of MUSIC to various perturbations is presented in [Rad94]. A compact expression for the error covariance of the MUSIC estimates may be used to evaluate the effect of various perturbation parameters including gain and phase errors, effect of mutual coupling, channel errors and random perturbation in sensor location. It should be noted that MUSIC estimates of DOA require knowledge of the number of sources, similar to certain other methods and underestimation of the source number may lead the inaccurate estimate of DOAs.

Analysis of the effect of model errors on the MUSIC resolution threshold and on the wave forms estimated using MUSIC indicate that the probability of resolution decrease with the error variance and that the sensitivity to phase errors depends more on array aperture than the number of elements in a linear array. The effect of gain and phase error on the mean source error (MSE) of the MUSIC estimates of the general array is analyzed in [4]. The problem of estimating gain and phase errors of sensors with known locations is considered in [14].

An analysis of ESPRIT under random sensor uncertainties suggests that the MUSIC estimates generally give lower MSEs than ESPRIT estimates. The former is more sensitive to the both sensor gain and phase errors, whereas the later depends only on phase errors. The study further suggests that for a linear array with a large number of elements, the MSE of the ESPRIT estimate with maximum overlapping subarrays is lower than nonoverlapping subarrays.

The effect of gain and phase errors on weighted Eigen space methods including MUSIC, minimum norm, FINE, and CLOSEST is studied by deriving bias and variance expression. This study indicates that the effects are gradual up to a point and then the increase in error magnitude causes the abrupt deterioration in bias and variance. The weighted Eigen space methods differ from the standard ones such that a weighting matrix is used in the estimate, and that matrix could be optimized to improve the quality of the estimate under particular perturbation conditions.

The effect of nonlinearity in the system on spectral estimation methods, including hard clipping common in digital beam former has been analyzed in [15]. It shows that by using additional preprocessing such distortion could be eliminated.

A summary of performance and sensitivity comparison of various DOA estimation schemes is provided in following tables

Table 3.1

Performance summary of Bartlett method

Property	Comments and comparison
Bias	Biased Bartlett>LP>MLM
Resolution	Depends on array aperture
Sensitivity	Robust to element position errors
Array	General Array

Table 3.2

Performance summary of MVDR Method

Property	Comment and Comparison
Bias	Unbiased
Variance	Minimum
Resolution	MVDR>Bartlett
	Does not have best resolution of any method
Array	General Array

Table 3.3

Performance summary of Maximum Entropy Method

Property	Comment and Comparison
Bias	Biased
Resolution	ME>MVDR>Bartlett
	Can resolve at lower SNR than Bartlett

Table 3.4

Performance summary of linear prediction Method

Property	Comment and Comparison
Bias	Unbiased
Resolution	LP>MVDR
	>Bartlett
	>ME
Performance	Good in low SNR conditions
	Applicable for correlated arrivals

Table 3.5

Performance summary of ML Method

Property	Comment and Comparison
Bias	Unbiased
	Less than LP, Bartlett, MUSIC
Variance	Less than MUSIC for small samples
	Asymptotically efficient for random signals
	Not efficient for finite samples
	Less efficient fro deterministic signals than random signals
	Asymptotically efficient for deterministic signal using very large array
Computation	Intensive with large samples
Performance	Same for deterministic and random signals for large arrays
	Applicable for correlated arrivals
	Works with one sample

Table 3.6

Performance summary of Element space MUSIC Method

Property	Comment and Comparison
Bias	Biased
Variance	Less than ESPRIT and TAM for large samples, minimum norm, Close to MLM, CLOSEST, FINE,Variance of weighted MUSIC is more than unweighted MUSIC Asymptotically efficient for large array
Resolution	Limited by bias
Array	Applicable for general array Increasing aperture makes it robust
Performance	Fails to resolve correlated sources
Computation	Intensive
Sensitivity	Array calibration is critical, sensitivity to phase error depends more on array aperture than number of elements, preprocessing can improve resolution Correct estimate of sources number is important. MSE depends on both gain and phase errors and is lower than for ESPRIT Increase in gain and phase errors beyond certain value causes an abrupt deterioration in bias and variance

Table 3.7

Performance summary of Beam space MUSIC Method

Property	Comment and Comparison
Bias	Less than element space MUSIC
Variance	Larger than element space MUSIC
RMS Error	Less than ESPRIT, minimum norm
Resolution	Similar to beam space minimum norm, CLOSEST Better than element space MUSIC, element space minimum norm Threshold SNR decreases as the separation between the sources increases
Computation	Less than element space MUSIC
Sensitivity	More robust than element space MUSIC

Table 3.8

Performance summary of Root- MUSIC Method

Property	Comment and Comparison
Variance	Less than root minimum norm, ESPRIT
Resolution	Beam space root MUSIC has better probability of resolution than beam space MUSIC
RMS Error	Less than LS ESPRIT
Array	Equispaced linear array
Performance	Better than spectral MUSIC Similar to TLS ESPRIT at SNR lower than MUSIC threshold. Beam space root-MUSIC is similar to element space root-MUSIC

Table 3.9

Performance summary of Minimum Norm Method

Property	Comment and Comparison
Bias	Less than MUSIC
Resolution	Better than CLOSEST, Element space MUSIC
Method	Equivalent to TLS

Table 3.10

Performance summary of CLOSEST Method

Property	Comment and Comparison
Variance	Similar to element space MUSIC
Resolution	Similar to beam space MUSIC Better than minimum norm
Performance	Good in clustered situation
Sensitivity	Increases in sensor gain and phase error beyond certain value causes an abrupt deterioration in bias and variance

Table 3.11

Performance summary of FINE Method

Property	Comment and Comparison
Bias	Less than MUSIC
Resolution	Better than MUSIC and minimum norm
Variance	Less than minimum norm
Performance	Good at low SNR

Table 3.12

Performance summary of ESPRIT Method

Property	Comment and Comparison
Bias	TLS ESPRIT unbiased LS ESPRIT biased
RMS Error	Less than minimum norm TLS similar to LS
Variance	Less than MUSIC for large sample and difference increases with number of elements in array
Computation	Less than MUSIC Beam space needs less computation than beam space root-MUSIC and ES ESPRIT Method LS ESPRIT is similar to TAM
Array	Needs doublets, no calibration needed
Performance	Optimum-weighted ESPRIT is better than uniform-weighted ESPRIT TLS ESPRIT is better than LS ESPRIT
Sensitivity	More robust than MUSIC and can not handle correlated sources MSE robust for sensor gain errors, MSE is lowest for maximum overlapping sub arrays under sensor perturbation

Chapter-4
Result and Analysis

4.1 Examples and discussion

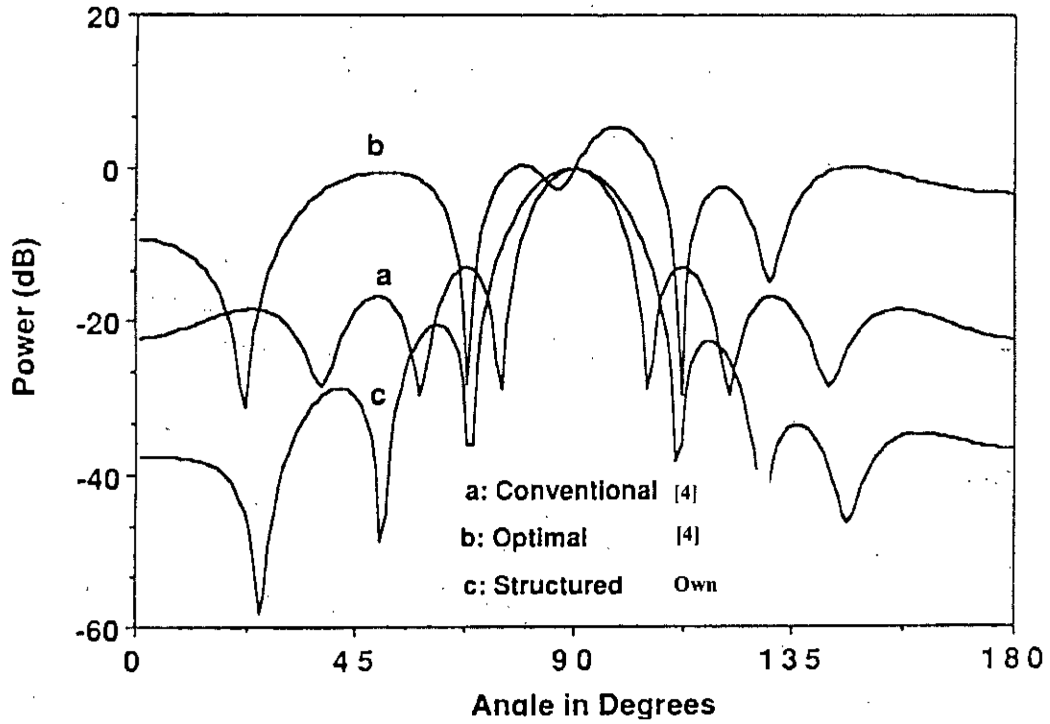

Fig 4.1 Power pattern of an element space processor using conventional, optimal and structured beam forming methods using an eight element linear array with one half wavelength spacing at the maximum frequency in the presence of six directional broadband interference in direction 22^0, 50^0, 68^0, 112^0, and 130^0. Look direction is 90^0 $\delta_n{}^2$=.0.01

Fig. 4.1 shows that power pattern of an eight- element linear array in the presence of six directional broadband sources using three beam forming methods. All sources are assumed to have the brick-wall type of spectrum with normalized cut-off frequencies of 0.45 and 0.5. The power of each sources 20 db above the power of white noise present on each element of the array. Five interferences are assumed to be in the far field of the array and are in the direction relative to the line of array and coincide with the side lobes of the conventional array pattern. The signal sources are to the array broadside. The interference in the direction 50^0 is fully correlated with the signal sources and delayed by 45^0 at the maximum frequency. The phase delay is specified at the origin of the co-ordinates system with array situated along the x-axis. The spacing between the elements of the array is taken to be one-half wavelength at the maximum frequency.

Fig 4.1 compares the power pattern of the conventional, optimal and structured beam formers. The figure shows that the power pattern of the optimal beamformar has an increased response in the direction of correlated jammer, and this increased response is responsible for the cancellation of look direction signal. The power pattern of the structured beam former shown in plot c has its response about -48 db in the direction of the correlated jammer and has clearly suppressed it. The plot of SNR measured at the output of the array using fig. 4.1 shown in fig 4.2 as a function of p_i/δ_n^2 for the plot c.

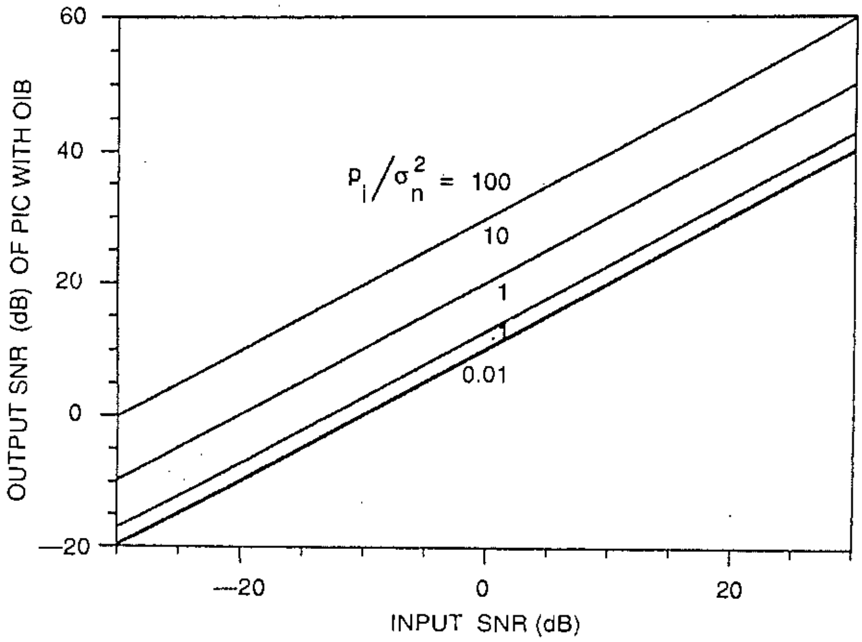

Fig 4.2 ***Output SNR of the postbeamformer interference canceller using orthogonal interference beam former vs. input SNR for eight-element linear array*** $\theta0=90^0$, $p1=1$, $\theta1=30^0$.

Fig. 4.2 shows output SNR vs. input SNR for various p_i/δ_n^2. The interference beam is formed using the steering vector in the end fire direction. For a given input SNR the output SNR increases as p_i/δ_n^2 increases.

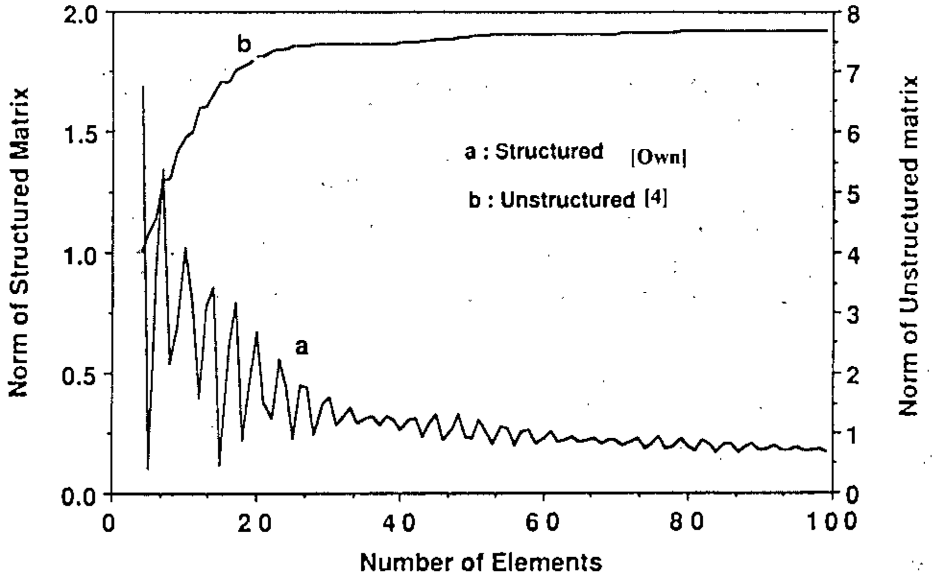

***Fig 4.3** Hilbert-Schmidt norm structured matrix vs. the number of elements in the array in the presence of one broadband correlated directional interference in direction 50^0 with unity power over frequency range (0.45, 0.5). The correlated phase delay is taken to be 45^0 at the maximum frequency measured at the origin. Look direction is 90^0, $\delta_n^2=0.01$.*

Fig. 4.3 compares the Hilbert-Schmidt norm of structured as well as the unstructured block of the array correlation matrix as a function of the number of elements in the array. The LXL dimensional block of the array correlation matrix considered corresponds to m=1 and n=1. Two sources are considered for the example. The look direction signal is broadside to the array and the correlated interference in the direction of 50^0 relative to the line of the array. The other parameters are the same as in the fig. 4.1. As seen in the fig.4.3, the norm of the structured correlation matrix decreases as the number of the elements in the array increases. On the other hand, the norm for the unstructured matrix [16] increases.

Chapter -5
Conclusion

5.1 Conclusion

A comparative study as well as an analytical analysis of different estimation techniques of signal arrival in a smart antenna has been carried out in this thesis. Some of the structured techniques have been compared with the conventional technique taken from [1]-[16]. It has been found that the structured estimation technique gives the better resolution, better sensitivity, and better performance than the conventional and optimal method.

Wireless communication has created a continuous demand for bandwidth increase and better quality of service. With the ever increasing number of mobile network subscribers, available capacity is becoming more of a premium. Smart antenna arrays are one way to accommodate this increasing demand for bandwidth and quality. These antenna arrays provide numerous benefits to service providers. However the processing requirement for smart antenna arrays is many orders of magnitude greater than those for single antenna implementations.

Smart antennas have the property of special filtering, which makes it possible to receive energy from a particular direction while simultaneously blocking it from other direction. This property makes smart antennas array effective tool in detecting and locating an underwater source of sound such as submarine without using active sonar. The capacity of smart antennas to direct transmitting energy toward a desired direction makes them useful for medical diagnostic purpose. The characteristic also make them useful in cancelling unwanted jamming signals. In communication system an unwanted jamming signal is produced by a transmitter in a direction of the desired signal.

The smart antenna field has been an active area of research for over four decades. During this time, many types of processors for smart antennas have been proposed and their performance has been studied. Practical use of smart antennas was limited due to excessive amounts of processing power required. This limitation has now been overcome to some extent due to availability of powerful computers and microprocessors.

Currently the use of smart antennas in mobile communication to increase the capacity of communication channel has reignited research and development in this very exciting field. Practicing engineers now want to learn about this subject in a big way.

The article aims to provide a comprehensive and detailed treatment of various antenna array processing schemes, adaptive algorithm to adjust the required weighting on antennas, direction of arrival (DOA) estimation methods including performance comparison.

Smart antennas involve processing of signals included on an array of sensors such as antennas, microphones and hydrophones. They have application on the areas of radar, sonar, medical imaging and communication.

5.2 Future works

Widespread interest in smart antennas has continued for several decades due to their use in numerous applications. The first issue of IEEE transactions of Antennas and Propagation, published in 1964 [IEE64], was followed by special issue of various journals [IEE76, IEE85, IEE87b], books [buc90, Mou84, Cad88] and a vast number of specialized research papers.

The current demand for smart antennas to increase channel capacity in the fast-growing area of mobile communication has reignited the research and development efforts in this area around the world. Research works can be initiated to evaluate the structured estimation techniques for signal arrival and its practical implementation in the smart antenna system. Design of parameters in smart antenna of various structured method can be initiated.

References:

[1] Smart antanns system- www.iec.org

[2] How to make an smart antenna- http://m.devlin@nallatech.com

[3] B.S, Guru and H.R, Hiziroglu, "Electromagnetic field theory fundamentals" PWS Publishing Company ITP An International Thomson Publishing Company Boston MA 022116-4324, 1998

[4] L.C. Godara, "Smart Antennas," CRC Press Inc.N.Y. 2004

[5] K.M. Buckley and X.L.Xu., "Spatial Spectrum Estimation in Allocation Sector," IEEE Trans on Accoust. Speech Signal Process vol.38, pp1842-1852, 1990.

[6] H.A, d'Assumpcao, "Some New Signal Processors for Array of Sensors", IEEE Trans.Inf.Theory vol. 26, pp.441-453, 1980

[7] H.A and G.E.Mountford, "An Overview of Signal Processing of Arrays of Receivers." IREE Aust, pp, 4-19, 1984

[8] B.D.Van Veen and K.M. Buckley, "Beamforming: A Versatile Approach to Spatial Filtering," IEEE ASSP Mag, pp.4-24, 1988.

[9] K.M. Buckley and X.L.Xu, "Spatial Spectrum Estimation in Allocation Sector," IEEE Trans on Accoust. Speech Signal Process vol.38, pp1842-1852, 1990.

[10] A.Cantoni, and L.C. Godara, "Resolving the Directions of Sources in a Correlated Field Incident on an Array", J. Accoust. Soc. Am., vol.67 pp.1247-1255, 1980.

[11]J.A cadzow, "A High Resolution Direction-of-Arrival Algorithm for Narrowband Coherent and Incoherent sources", IEEE Trans. Acoust. Speech signal process, 36,965-979, 1988

[12] L.C Gadara, "Limitations and capabilities of Direction –of –arrival estimation techniques using an array of antennas: A mobile communication perspective", paper presented at IEEE international symposium on phased array systems and Technology, Boston, 1996.

[13]W Du, and R.L Kirlin, "Improved spatial smoothing techniques for DOA estimation of coherent signals", IEEE Trans, signal process, 39, 1891-1892,1991.

[14] Y. Meng, P. Stocia, and K.M. Wong, "Estimation of the direction of arrival of spatially dispersed signals in array processing", IEE proc. Radar sonar navig,143 1-9, 1996

[15] B. Ottersten, and T. Kialath," Direction of Arrival Estimation for Wide Band Signal Using the ESPRIT algorithm," IEEE Trans. Acoust. Speech signal process, 38,317, 1990

[16] Nagasuka, M, et al., "Adaptive array antenna based on spatial spectral estimation using maximum entropy method," IEICE Trans commu, E77-B 624-633, 1994