Technical Report

Investigation of Algorithms for a Spatially Adaptive Antenna

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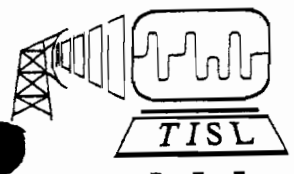

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Purpose: This is an interim technical report of the work, progress, and accomplishments for the investigation of algorithms to control a spatially adaptive antenna array. This report was prepared according to following:

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Contract: Investigation of Algorithms for a Spatially Adaptive Antenna

Contracting Agency: RADC/DCCL
Griffiss Air Force Base
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Scope: This report covers the items shown in Statement of Work as listed below:

Task 1: Review of HF Environment
HF Channel Model
Channel Simulation

Task 2: Review of HF Antenna Arrays
HF Array Model
Review of Adaptive Algorithms
Initial Simulation

ABSTRACT

The initial investigation of algorithms to control a spatially adaptive planar antenna array is described in this report. The report provides details of work in three areas: HF Environment; Planar Arrays; and Adaptive Algorithms.

The work in the HF Environment was to investigate and summarize key findings in open literature in the area of HF channels, HF systems, and HF channel models. A summary description of the physical HF channel and HF channel models is presented. An HF channel model was constructed using a computer program. It was tested and demonstrated to be a satisfactory model. It is based on the Signatron model (13) and the original L. Ehrman model (22). The channel model is used in conjunction with an antenna model and adaptive algorithm to simulate and operate an adaptive array.

A planar array model was constructed using a computer program. It represents an arbitrary shaped, two dimensional, planar array of m x n elements. The array is completely controlled in amplitude and phase to allow beam steering, sidelobe control, and placing of nulls. The array used with the HF channel model serves as a test bed for the adaptive algorithm.

A model of an adaptive system was defined. It serves as a basis for initial investigation and characterization of adaptive algorithms. Both recursive and iterative algorithms are defined, described and characterized in the report. A simulation was performed using the Least Mean Squares (LMS) algorithm. The LMS appears to be the most promising algorithm to use at this time.

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1.0 INTRODUCTION

The initial investigation of an algorithm for a spatially adaptive antenna and the work done to date is described in this Interim Technical Report. The work and topics are as follows:

Task 1: Review of HF Environment

- 1.1 Review and Summary of HF Channels
- 1.2 HF Channel Model
- 1.3 HF Channel Algorithm
- 1.4 Simulation
- 1.5 Review and Summary of Task 1

Task 2.1: Review of Arrays and Algorithms

- 2.1.1 HF Antenna Arrays
- 2.1.2 HF Antenna Model
- 2.1.3 HF Array Algorithm
- 2.1.4 Simulation

Task 2.2: Processing Algorithms

- 2.2.1 Overview of Adaptive Algorithms
- 2.2.2 Adaptive Algorithm Model and Parameters
- 2.2.3 Comparison and Analysis of Adaptive Algorithms

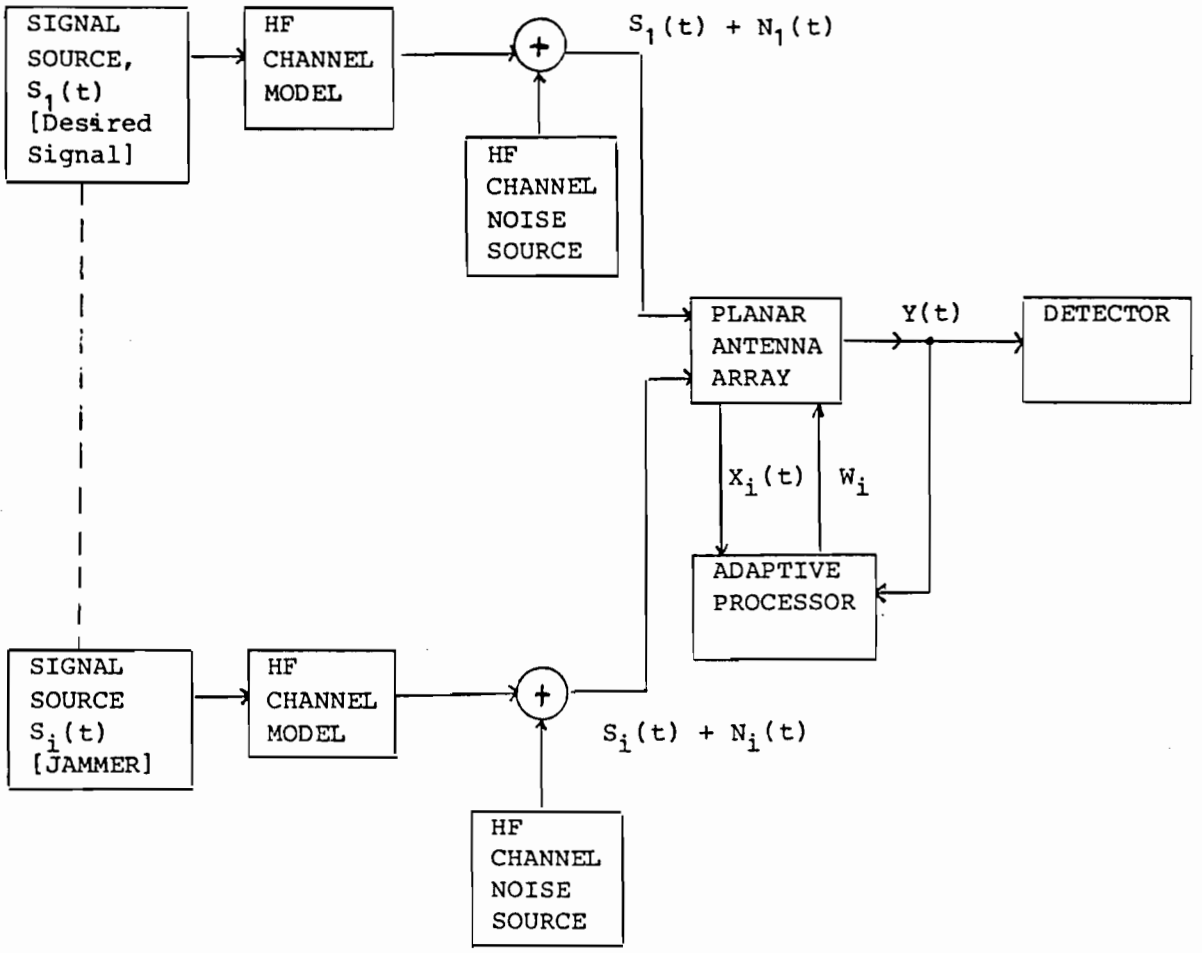
- 2.3 Review and Summary of Task 2

Our investigation of algorithms to control an adaptive HF antenna array began with the basics of the HF channel, HF arrays, and a review of adaptive algorithms. Included in this preliminary phase was development of a simulation system containing a planar antenna array and an HF channel model. This simulation system will be used to test and evaluate candidate adaptive algorithms. Thus adaptive algorithms will be studied, tested and evaluated in the system environment where they will be used. This simulation system includes a signal generator, a tapped delay line channel model, planar antenna array, and

the adaptive processor. This is shown as a system model in figure 1.0-1. This system model will serve as a reference for this report.

The system model shown in figure 1.0-1 is a typical HF system containing all the system elements including jammers. By first developing models of the various parts of the system we were able to provide a realistic test and evaluation environment for our algorithm investigation. In addition to serving as an example of a typical system, the model in figure 1.0-1 also illustrates the task breakdown for our investigation and the limits and interface of each block in the model.

In this Interim Technical Report we will describe the channel, antenna, introduction to candidate algorithms, and discuss how they are all integrated into one system. The Final Technical Report will contain details of algorithm evaluation, testing, and hardware implementation. The software system for candidate algorithms will be provided as part of Task 4 but as a separate deliverable item.



$S_1(t)$: Desired input signal
$N_1(t)$: HF channel noise associated with $S_1(t)$
$S_i(t)$: Typically a jammer or other interference
$N_i(t)$: HF channel noise associated with $S_i(t)$
$X_i(t)$: Composite signal from each element in the antenna array
W_i: Antenna element control weight
Y(t): Composite output of antenna array

Figure 1.0-1 HF System Model

Task 1

Review of HF Environment

1.1 Review of literature and HF channel characterization

1.2 HF channel model block diagram, characterization and parameters

1.3 HF channel model algorithm, software and model simulation

1.4 Simulation and results of HF channel model

1.5 Review, summary and report of Task 1 results

1.1 Literature Review and HF Channel Characterization

1.1.1 Literature Review

The objective of the literature review was to investigate and summarize key findings from published papers, articles, books, reports and other special publications in the area of HF channels and systems. The findings are generally known by most people with a serious interest in HF and who are currently working in this area. However, it is useful to categorize these findings in the beginning of the program for all to read. The result is not intended to be a tutorial on HF but rather a summary description of milestone papers, and key findings that affect implementation of a HF channel model. The following is a summary of articles, books, conference proceddings and special issues that are either milestone events in HF, or provide some tutorial material in some aspect of HF. Citations are for both HF channel characterization and HF channel models.

Books and Reports

1. Kennedy, R. S., Fading Dispersive Communications Channels, Wiley Interscience, 1969.

   Text with complete treatment of fading dispersive channels. The results are relevant to HF. However, the scope of the text is much broader than HF and is applicable to all fading dispersive communication channels. Material is mostly theoretical, with few practical examples. Included is a general channel model, general channel characteristics, system modules, error performance, waveforms and eigenvalues, and demodulators for fading dispersive channels.

2. American Radio Relay League Staff, The Radio Amateur's Handbook, American Radio Relay League, Newington, Connecticut 06111, 1981.

   General handbook on amateur radio equipment and techniques. Chapter 18 however, is dedicated to an explanation of short wave propagation, as it is related to amateur radio. The book describes the influence of solar

activity, ionozation layers, propagation modes, angles, distances, and propagation predictions. Chapter 20 on HF antennas is of related interest with practical details on theory, implementation, and use of HF antennas.

3. Dolukhanov, M. V., Propagation of Radio Waves, MIR Publishers, Moscow, 1971 (translated from the original Russian).

A complete text on wave propagation for systems in the bands from VHF and below, down to very long wave frequencies. A major portion of the text is about the troposphere, ionosphere, and propagation through these portions of the atmosphere. Covers all major aspects of HF propagation, except does not include any models. An excellent reference.

4. Orr, William, Radio Handbook, 22nd Edition, Howard W. Sams Publishers, 1984.

Covers all major aspects of radio equipment, operation, transmission, and reception. Chapter 23 is solely on propagation. Included is the influence of solar activity and the resulting ionospheric layers, frequency predictions, and general propagation predictions. Related material is presented on sunspot activity, ionospheric disturbances, and radio noise in the HF band.

5. Brayer, Kenneth, Editor, Data Communications via Fading Channels, IEEE Press, 1975.

A collection of articles in three areas: Fading Channels, Digital Signalling on the Fading Channel, and Error Control for the Fading Channel. There are many articles of general interest and usefulness concerning the HF channel and HF transmission. Two classic landmark papers are presented: Bullington (Radio Propagation Fundamentals, BSTJ, Vol. 36, May 1957 pp 593-625), and McNichol (The Fading of Radio Waves of Medium and High Frequencies, proceedings of the IEEE, Vol. 96, Part III, November 1949, pp. 517-524). This collection on data over fading channels contains many basic and tutorial concepts regarding the fading dispersive channel.

6. CCIR, Handbook on High Frequency Directional Antennae, International Telecommunication Union, Geneva, 1966.

Handbook mostly concerned with HF antennas and arrays. However, some material is included regarding angle of arrival of a signal as related to the path distance and varying heights of the ionosphere. Lot of practical results from HF systems operated by various PTT administrations which belong to the ITU.

7. CCIR, Fixed Service at Frequencies Below about 30, MHz, Volume III, Geneva, 1982.

CCIR recommendations for point to point service in the HF band. Useful details on noise levels, interference, antennas, frequencies, propagation, and coordination. Mostly written for the administrations but good reference for anyone operating links in this service.

8. CCIR, Propagation in Ionized media, Volume VI, International Telecommunication Union, Geneva, 1982.

Companion publication to Volume III, Fixed Service, but emphasizing short wave propagation and other propagation due to ionized layers in the atmosphere.

9. Davies, K., Ionospheric Radio Waves, Blaisdell Publishing, Ginn and Company, Waltham, Massachusetts, 1969.

Textbook on ionospheric propagation based on lectures given at the University of Colorado and Institute for Telecommunication Science, Boulder, Colorado. Provides a complete treatment of ionospheric propagation covering: E & M theory, ionospheric structure, waves in ionized media, dispersion, absorption, ray paths, layer heights, variation in phase and frequency, whistlers, magnetoionic effects, oblique propagation, fading sounding, and unusual nonlinear processes.

10. Folkestad, K., Editor, Ionospheric Radio Communications, Plenum press, New York, 1968, Conference Proceedings of a NATO Conference in Norway, 13-19 April 1967, on Ionospheric Radio Communications.

The proceedings include papers presented in the areas of ionospheric physics, system and equipment design, and system applications. The emphasis of the conference was on high latitude (Artic) propagation and systems. Special topics on high latitude propagation presented were: ionospheric and meteorscatter propagation, abnormal auroral absorption, polar cap absorption, electron density in high latitude regions, HF predictions, fading in the auroral regions, system problems and unique problems in the Artic region.

11. Collins Telecommunications Products Division, HF Communications Data Book, Rockwell International, Cedar Rapids, Iowa.

A complete handbook on HF transmission, propagation, and reception. Subjects included in propagation are: sky wave, ground wave, frequency selection for MUF and FOT, frequency and propagation charts, ionospheric disturbances, magnetic storms, and a listing of forecasting services. Topics included under transmission and reception relating to the antenna are: polarization choice and efects, ground effect on pattern, radiation angle, gain, and radiation patterns.

12. Hunsucker, R. D., High Frequency Propagation Prediction and Analysis for Circuits from the USCG San Francisco Radio Station to Ships and Aircraft operating in the North Pacific, Report OT TRER 15, U.S. Department of Commerce, Office of Telecommunications, Washington, D. C. 20005, 1971.

Investigation of the general high latitude HF propagation problem using a fixed station in San Francisco and several ships and aircraft operating in the North Pacific. The report provides models and details of the expected path loss at high latitudes due to : Non-Great-Circle propagation, D region absorption in auroral zones, sporadic E effects, F1 layer effects, and other anomalous effects. Has a good summary of overall results of propagation reliability. The report comments on the ITS propagation

prediction program and changes required to include the effects of high latitude transmission.

13. Signatron, High Frequency Channel Simulator, Technical order, Signatron, Lexington, Massachusetts 02173, December 1979.

A technical order (report) containing all the details needed to maintain the hardware and software of the HF Media Simulator. Gives details on the hf channel simulator designed by Signatron and the necessary details to operate the simulation sytem. Of particular interest is the channel model used in the simulator and its similarity to the models described in most references on HF channel models and the HF model used in this study of adaptive algorithms.

14. Modestino, J. W., Jung, K. Y., Matix, K. R., and Vickers, A. L., Design and Implementation of the Interactive Communications Simulator (ICS), Final Technical Report, RADC-TR-81-37, April 1981, Rome Air Development Center, Air Force Systems Command, Griffiss Air Force Base, New York 13441.

A report on a comprehensive system for the digital simulation of a wide variety of point to point communication systems. Of particular interest is the channel model used for the HF channel. The particular channel model in ICS confirms the use of the same model for the study of an adaptive algorithm, as described in this report.

15. CCIR Draft Report 549-1, HF Ionospheric Channel Simulators, CCIR, Volume III, CCIR XIV plenary Assembly, Kyoto, 1978, international Telecommunication union, Geneva, 1978.

A report on a proposed HF channel model, as developed by C. Waterson. The model simulates the short term distortion effects of the channel due to multi-path and multi-mode propagation, as in the Signatron model, and shown by L. Ehrman.

Conference Publications

1. AGARD Confeence Proceedings Number 173 on Radio Systems and the Ionosphere, NATO, AGARD, Specialist Meeting of the Electromagnetic Panel held in Athens, Greece, 26-30 May 1975.

   Conference publication with complete copies of 31 papers presented at the conference. Topics presented were in the areas of: 1. Satellite Systems: scintillation effects; and satellite ionospheric effects on navigation systems. 2. HF Systems: sounding and predictions; optimum channel selection; system design parameters; high latitude systems; and DF systems. 3. LF/VLF Systems. Papers included current work in HF data transmission, channel sounding , and channel characterization and prediction.

2. IEE Second Conference on HF Communication Systems and Techniques, IEE, London, 15-17 February 1982.

   Collection of 29 papers given at the IEE Conference in London, February 1982. Topics are grouped into: Systems, Real Time Evaluation and Simulation, propagation and Interference Analysis and modeling, and Signal Generation and Processing.

Special Issues and Special Publications

1. Joint Technical Advisory Committee (JTAC), Radio Transmission by Ionospheric and Tropospheric Scatter, Proceedings of the IRE, January 1960.

   A landmark issue of the Proceedings of the IRE regarding scatter propagation. Ionospheric scatter systems are described in great detail. Ionospheric scatter systems as described in this issue refer to systems using D region scattering and operating in the 35-50 MHz band just above the HF band. The article presents a comparison between the D region scatter circuits and typical HF circuits. Included is an extensive bibliography of significant references on scatter communication principles and systems.

2. Goldberg, G., Communication Channels: Characterization and Behavior, IEEE Press, New York, 1976.

Collection of tutorial articles on propagation in the bands from VLF, LF, MF, HF, VHF, UHF, microwave, millimeter, to optical. There is one complete section on HF which includes: analytical channel model, channel transfer functions, channel simulation concepts, comparative performance of systems on actual channels, and error rate behavior. At the time of publishing the collection was fairly complete in these subject areas.

Articles

1. Damboldt, I., A comparison Between the Deutsche Bundespost Ionospheric HF Radio Propagation Predictions and Measured Field Strength, AGARD Conference Proceedings 173, Athens, May 1975.

Report on a comparison between long term forecasts and actual measured field strength. The paths measured were for transmitters in Tokyo, Halifax, Mauritius, Canberra, and Bracknell, England; all to a receiver in one location in Germany. The propagation predictions are for long term median field strength as a function of frequency in the HF band. Median values are predicted by a formula from Beckman, 1965, from a German publication (title and translation not available). Main value is the relationships from Beckman which show the prediction for median field strength for 2,000-4,000 km paths.

2. Dawson, J., An HF Simulator for use with Real Time Channel Evaluation Systems, IEE Second Conference on Hf Communication Systems and Techniques, London, February 1982.

An Example of a real time HF channel simulator based on the Waterson model. The model was used as a test bed for off air evaluation of an HF radio link. Analogous to the channel simulator implemented at RADC for modem testing but using different techniques.

3. Ehrman, L., Bates, l., Eschle, J., and Kates, J., Real-Time Software Simulation of the HF Radio Channel, IEEE Transactions on Communications, Vol. COM-30, No. 8, August 1982, pp. 1809-1817.

Landmark paper used as a reference for the tapped delay line model of the HF channel. The specifications are according to the details proposed by Waterson (Waterson, C. C., et al., An Ionospheric Channel Simulator, U. S. Department of Commerce, ESSA Research Lab, Boulder, Colorado, Technical Report ERL TM-ITS 198, September 1968).

4. Elvy, S., A Design for an Automatic HF Radio System, Radio Science, Vol. 20, No. 3, May-June 1985, pp. 261-268.

Report on the design of an HF radio terminal which could be used for low speed data or duplex voice. The design makes use of the latest electronic techniques and covers operating protocols, message formats, and frequency selection based on ionospheric data and channel occupancy information.

5. Goldberg, B., 300 KHz-30 MHz MF HF, IEEE Transactions on Communication Technology, Vol. COM-14, December 1966, pp. 767-784.

A landmark paper on MF and HF propagation. Contains useful tutorial material on propagation in these bands. Included are topics on signal levels and noise, channel transfer properties, interference, fine grain behavior, and system performance.

6. Hunsucker, R., Survey of Polar and Auroral Region Effects on HF Propagation, Radio Science, Vol. 4, No. 4, April 1969, pp. 347-365.

Report on the effect of the auroral and polar zone ionosphere on sky wave propagation in the HF band. Included is an explanation and examples of: non-great circle mode of propagation, sporadic E ionization, F1 layer effects, F region irregularities, D region absorption (Polar Cap absorption), and reflectivity properties of the polar ice caps. Also reported is field data taken during tests at the University of Alaska. This report is the most up to date and complete found on Polar Cap and Auroral absorption.

7. Ince, A. N., Communications Through EM-Wave Scattering, IEEE Communications Magazine, May 1982, pp. 27-43.

A tutorial article on propagation due to scattering from inhomogeneities such as meteor trails, and the troposphere, as distinguished from the usual HF upper ionospheric reflection and refraction of the radio waves. The article is relevant to HF propagation in that under certain conditions the usual upper ionospheric HF path is replaced by a lower level scatter path in the D region. In addition a lot of the characteristics of a true scatter path are similar to the usual upper ionospheric reflecting HF path.

8. Perry, B. D., A New Wideband HF Technique for MHz-Bandwidth Spread-Spectrum Radio Communications, IEEE Communications Magazine, September 1983, pp. 28-36.

A discussion of high data rates for HF paths. Includes relevant material on propagation and multipath which is applicable to all HF paths.

9. Wagner, L. S., and Goldstein, J. A., High Resolution Probing of the HF Ionospheric Skywave Channel: F2 Layer Results, Radio Science, Vol. 20, No. 3, May-June 1985, pp. 287-302.

An excellent tutorial and general interest article on HF channel sounding. Shows data and actual results of channel sounding.

10. Waterson, C. C., Juroshek, J. R., and Bensema, W. D., Experimental Confirmation of an HF Channel Model, IEEE Transactions on Communication Technology, Vol. COM-18, No. 6, December 1970, pp. 792-803.

A landmark reference in the development of the tapped delay line model of the HF channel. This article gives all the details of the makeup and parameters of the tapped delay line model. Also included is a comparison between results from the model and actual path data. The development of the model is related to the authors work for the U. S. Department of

Commerce, ESSA, Boulder, Colorado. (Same authors, Experimental Verification of an Ionospheric Channel Model, ESSA, Boulder, Colorado, Technical Report, ERL-112-ITS 80, July 1969).

11. Day, W. E., Meteor Burst Communications Bounce Signals Between Remote Sites, Electronics, 29 December 1982, pp. 71-75.

An article on the practical use of meteor trails and a brief explanation of the scatter mechanism in the D and E region.

12. Modestino, J. W., and Matis, K. R., Interactive Simulation of Digital Communication Systems, IEEE Journal on Selected Areas in Communications Vol. SAC-2, No. 1, January 1984, pp. 51-76.

A thorough explanation of the use of the ICS model for channel simulation. An excellent description is given of the ICS model including all the model elements and parameters. Examples of simulation runs are provided.

1.1.2 HF Channel Characterization

An HF channel model has evolved over the past fifteen years, beginning with the Waterson model (22, 29). The basis for this model and other models (13, 14, 15, 21) has been the characteristics of the HF channel. To provide insight into the HF channel model and to establish some definitions and assumptions made in our study, it is valuable to review and state the key findings relative to HF propagation that are incorporated in the model and used in our study. These details are from the publications reviewed in Section 1.1.1. Literature Review and cited in the list of references at the end of this report. These details are all generally accepted and used for any HF channel simulation and HF systems engineering. Details given below also serve as a reference for any assumptions made in our study regarding HF propagation. Review of these details will allow one to quickly assess the validity of our assumptions; our model and baseline for HF channel simulation; and use of the model for investigation of an adaptive algorithm.

The characteristics of the HF channel that are modeled for use in investigating an adaptive algorithm are the short term fading and dispersive effects of the HF sky wave path. The short term fading and dispersive phenomena are the principle effects modeled and are shown by use of a tapped delay line model, as developed by Waterson (29). The channel characteristics result from many natural phenomena which are reviewed below. They are in the following areas:

Ionospheric physics
Radio wave absorption anomalies
Radio noise peculiar to the HF band
Propagation modes
Short term loss factors
Dispersive characteristics

1.1.2.1 Ionosphere

The structure of the ionosphere, different layers, ion density, and heights of the layers are central to HF sky wave propagation. This has been reported extensively and is the subject of most of the references cited. The general results are summarized below regarding ionospheric characteristics that affect the HF sky wave.

D Region: Exists only in the daytime, is between 50-90 km above the earth and causes little if any reflection of the HF radio wave. Major effect of the D region is absorption due to ionic collision. As the absorption increases as the frequency decreases (and the radio noise increases as the frequency decreases) then the main constraint introduced is a lower limit on the operating frequency. The lower limit is referred to as the LUF (lowest useable frequency).

E Region: Also only exists during daylight hours. This region is found from 90-140 km above the earth. Main effect of the E layer is also absorption, but to a lesser extent than the D layer. However, an additional effect in this layer is sporadic E which is responsible for scatter propagation. Scatter propagation

caused by sporadic E depends on the latitude (above or below the equator) and is the result of reflections of the sky wave in this ionic layer. This is caused by irregularities in ion density due to turbulence and from meteor trails (18, 19, 25, 26, 30).

F1 Region: Similar to the E region during daylight and exists at a height of 140-250 km above the earth. During nighttime this region rises to merge with the F2 region. (Meaning that there is no distinguishable F1 layer at nighttime.)

F2 Region: The highest ionospheric layer, from 150-250 km above the earth during nighttime and rises to 300 km during daylight. This region is the most useful for skywave transmission as it is the F2 layer that provides the refraction necessary to bend the radio wave back to the earth for long distance communications. As the refraction process is a function of the refractive index which depends on the ion density and the frequency, then it is the F2 layer which sets an upper limit on the operating frequency. Frequencies above a maximum useable frequency (MUF), will penetrate through the F2 layer and can not be received. Frequencies below the MUF will be reflected.

Notes: The ionospheric regions and the effects of these regions vary greatly depending on the latitude and longitude of the mid-region of the path. Generally, the biggest changes (and the most detrimental to propagation) take place in the high latitudes. The high latitudes generally have two regions of anomalous propagation: Auroral zones from 60-80 degrees North (10) and Polar Cap from 70-90 degrees North (25). These effects are described in the following section on natural phenomena.

1.1.2.2 Natural Phenomena

Natural phenomena are classified as naturally occurring phenomena which cause anomalous propagation. The phenomena of interest are the ones which

cause high absorption and can render a sky wave path completely unuseable for minutes up to days at a time. These phenomena vary greatly depending on location and must be accommodated in the system design.

Solar Activity:

As the sun is the only source of energy for ionization of the atmosphere then any variations in the sun's intensity or solar disturbances have an effect on the ionized regions which control sky wave propagation. The major variations which noticeably effect HF sky wave propagation are: the eleven year cyclic sunspot activity, solar eruptions, and particle radiation. The main effect is a variation in ionic density which changes any or all of the: MUF, D and E region absorption and the LUF. In any case these are long term effects and are to be included in the long term path loss.

Sudden Ionospheric Disturbances (SID):

Occasionally during normal daytime communications, the HF sky wave is greatly attenuated causing a complete loss of communications. This is generally caused by an SID due to a solar eruption. The solar eruption creates a great increase in ion density in the D and E regions and hence greatly increases the absorption in these layers. Due to the increased ion density the LUF is effectively lowered for the path. The SID can be treated as a long term power fade as it is relatively steady and can last anywhere from ten minutes to several hours.

Magnetic Storms:

Effects of a magnetic storm are similar to the effects of an SID; long term power fading. However, the storm and the SID are different phenomena, though they both originate from the sun. Magnetic storms are believed to be caused by particle radiation from the sun. Effects are twofold: the F2 layer ion density is reduced which reduces the MUF, and the D and E region absorption is increased. A magnetic storm may last for several days. It has a very different time statistic compared to the SID.

Auroral Zone Absorption:

Increased absorption in the auroral zones (60-80 degrees North) is produced by electrons with unusually high energy levels (25). The ionization varies widely depending on time of day and location. Regarding the HF channel model, the increased path attenuation in the auroral zones is considered to be a long term power fade and is not considered a parameter of the channel model.

Polar Cap Absorption:

This is also increased absorption, however, it is restricted to paths which traverse the polar cap region (70-90 degrees North). The statistics of polar cap absorption are different from auroral zone in that the effect is uniform throughout the area, it is relatively infrequent, causes almost total absorption, and the effect may last from several days to a week (16). This is also treated as a long term power fade and not included as an HF channel model paramter.

Non Great Circle Propagation (NGC):

At high latitudes an HF sky wave may deviate laterally from the great circle path (it would normally follow) by tens of degrees (12, 25). The overall effect is a drop in signal level and excessive distortion due to the radio path geometry changing from what one expects and desires. As this is a long term loss effect it is not included as a parameter in the HF channel model.

Magnetoionic Effect (Faraday Rotation):

Due to the earth's magnetic field a linearly polarized wave is divided into two orthognally polarized waves, sometimes referred to as an ordinary and extraordinary ray (3). The polarization is dependent on the frequency and the original polarization of the transmitted wave. For the sky waves in the HF band, for linearly polarized transmitted waves, the received waves are generally eliptically polarized (3). The resulting effect at the receiver is a 3dB loss in signal level due to the antenna being linearly polarized. This is a long term effect and is included in the long term median received signal

level. It is not included in the HF channel model as it does not effect the simulation for purposes of the adaptive processor for an HF antenna array.

1.1.2.3 Radio Noise

To achieve reliable communications, the minimum signal level required (to operate above the noise) is a function of the ambient noise level. This is the usual constraint on all communication systems. However, there is a difference in the ambient noise source and level if one is in the HF band (3-30 MHz) and below or above the HF band. Above the HF band the ambient noise is generally a function of the receiver noise temperature or other noise associated with the receiver system.

In the HF band and below reliable communications is limited by external noise sources. This creates a severe problem for the HF communicator as the external noise sources, levels, and statistics, vary greatly depending on environment (rural vs. urban), time of day, location of mid-path (latitude and longitude), and frequency of operation. Therefore the HF channel models include a provision for adding noise as appropriate and desired, to the signal when using the model for simulation. However, the noise sources are not included as an inherent feature of the channel model. The sources, such as Gaussian or impulsive, are external to the channel model. (Recall that the model is a tapped delay line to simulate phase interference fading independent of ambient noise levels or statistics.)

Due to the variations in external noise with respect to level, statistics, and frequency, then preprogrammed noise sources are added to the signal, independent of HF channel multipath fading. External noise sources which affect HF circuits may be classified into two general areas: Random and Impulsive.

Type	Atmospheric Sources	Equipment Sources
Random	Distant Lightning Cosmic RF Interference	Intrinsic Receiver Noise
Impulsive	Local Lightning Man Made RF Interference	Power

In general, with currently available receivers, the main noise contributors are atmospheric (in rural areas), and both atmospheric and man made in urban areas. In these areas the main contributions are from lightning, both impulsive and random, and usually greatly exceed any cosmic or equipment thermal noise.

1.1.2.4 HF Propagation

Radio Path

In the HF band (as in most other radio bands) a radio path may be established either as a ground wave (point to point along the surface of the earth) or as a sky wave via reflections from layers in the atmosphere or diffuse scattering from areas in the atmosphere (such as troposcatter or meteor scatter). In HF the ground wave can be used for distances up to about 100 Km depending on frequency, terrain, and location. The sky wave can be used for paths from about 150 Km and greater depending on whether the path is a reflecting or a scatter path. In certain cases a sky wave can be received from two opposite directions, the longer path being from the reverse direction around the earth. For our investigation and for the HF channel model employed here we are concerned with the reflecting sky wave only, received via multiple modes and reflection mechanisms. (Mode as used here refers to the number of major resolveable paths the signal may take between transmitter and receiver.)

The long distance HF sky wave can be received via several different paths and mechanisms. Given an optimum working frequency and properly aligned antennas a sky wave path is usually set up for a single hop from an F2 layer

reflection. This is useable out to about 4,000 Km. Multiple hops can theoretically (and practically) propagate around the earth. In addition to single or multiple hop F2 layer reflections, a long distance signal can also be received via sporadic E scatterers. However, this is normally a much weaker signal then the F2 reflected signal and is not significant unless the F2 path is greatly attenuated, such as operating above the MUF for current F2 conditions.

The propagation conditions described above are characteristic of almost all useable HF sky wave channels. For these channels propagation may be categorized as either an F2 reflection channel with a single or multiple hop and/or a scatter channel. For an F2 reflection channel the number of hops supported is defined as the number of modes of propagation. (Ideally, one wishes to receive only one mode.) For either a single F2 hop or a scatter channel the link is characterized as supporting a single mode. Higher modes are categorized by the number of useable paths which can be different modes by the tapped delay line which is the basis for the model. For purposes of this report, an individual channel mode is analagous to a tap in the tapped delay line. The mode (tap) parameters which are under control of the user are as follows: number of modes (usually taken as 3); time delay between modes; time spread within each mode; and the relative amplitude of the signal via each mode.

Loss Factors

For sky wave propagation the median path attenuation is principally caused by beam spreading (as shown by Maxwell's equations), absorption in the D, E, or F1 layers, polarization loss, and ground reflection loss for multiple hop paths. These are long term loss factors which have both a diurnal (daily) variation, and a seasonal and cyclic (11 year sunspot) variation. These loss factors are to be distinguished from the short term loss in a path caused principally by phase interference fading (9). The long term, median, loss factors are not included in the path loss model. They are specified as signal parameters to be determined by the model user when defining the signal source and noise levels. The distribution usually assumed for the long term median loss is the log-normal distribution.

Short term loss (phase interference fading) is caused by slight changes in path distance within a particular mode or between modes (multimode reception). This is accommodated in the channel model and is the principle fading effect exhibited by the model. The short term fading distribution is the Rayleigh distribution. For the case of an F2 reflection channel and a superimposed scatter channel, the short term distribution has a different shape and is characterized by the Rice distribution.

Doppler Shift

Due to turbulence in the F2 reflecting layers or the sporadic E layers, doppler shift may be introduced in the carrier frequency. For an AM-SSB typically used in HF this translates to an equivalent shift in the frequency in the baseband. However, for most paths the doppler shift rarely exceeds a few Hz (9). Due to the small doppler shift and that there is little if any effect on the channel simulation as it affects the adaptive processor for an antenna array, then doppler shift is not included in the model when used for the antenna array processor.

Delay Spread

A pulse transmitted via an HF sky wave will be received as a series of pulses spread out in time. The time spread being a function of the power received via different ionospheric paths of different lengths. Within a signal mode or between modes the time delay difference is referred to as the delay spread. As the reflecting layers of the ionosphere and the sporadic E layers can have significant vertical dimensions, then there can be significant delay spread within a mode and between modes. Delay spread is an important feature of the HF channel model and is under control of the user to simulate various single hop and multiple hop channels.

The delay spread within a mode and between modes varies considerably depending on the path, frequency, antenna pattern, and propagation conditions. However, there is generally a range of delays which have been found to be typical of most HF sky wave paths:

Within a mode: from 100 microseconds to 10 milliseconds with typical being on the order of 100-500 microseconds (19).

Between modes: time delay differences depending on path length differences at the rate of 3 microseconds difference per Km difference in path length. The biggest differences are for short paths. For long paths the multimode differences are not any longer than the delay spread within a single mode.

An example of time delay difference is shown below. For the HF band of frequencies with a wavelength of 100 meters to 10 meters the time delays for all cases amount to many cycles of the radio carrier, hence the Rayleigh distribution.

Mode	Path Distance, Km		Absolute Mean	Estimated Max.
	Ground	Skywave	Delay, msec	Delay Spread, msec
1F2	500	673	2.3	1.4
	1000	1097	3.7	1.4
	2000	2050	6.8	1.4
	4000	4025	13.4	1.4
2F2	500	1030	3.4	2.8
	1000	1345	4.5	2.8
	2000	2193	7.3	2.8
	4000	4100	13.7	2.8

Note: 1F2 refers to one hop via the F2 region.
2F2 refers to two hops via the F2 region.

1.1.2.5 Path Characterization Summary

For investigation of an algorithm for an adaptive HF antenna array, one is only concerned with characteristics of the HF channel that are relevant to the investigation of the array control requirements. There are many characteristics that could be included in the algorithm study and implemented in an HF channel model. However, it was found that a limited number of characteristics would be adequate without limiting the scope of the investigation. The main channel characteristics that are implemented are the short term (phase interference) fading and the delay structure of the individual modes (paths). This is the same as the models described by Signatron (13) and ICS (14), with the exception of the following items which are not included in our model:

1. Doppler component due to ionospheric turbulence is not necessary as it does not affect array control.

2. Only one magnetoionic component is included rather than two. A second component does not affect array control.

The two main phenomena, fading and delay, included in the model, are summarized below.

Fading

The main purpose of the channel model is to simulate short term fading. (Short term is interpreted as changes that take place within a minute or less. By comparison long term is used to denote changes in path loss over periods of an hour.) For the HF path short term fading is predominantly phase interference fading. The mean attenuation of the HF path does not change significantly in a short time period (within a minute) and is considered as a long term characteristic. Phase interference fading as implemented in the model is due to time differences in signal paths both within a single mode, between modes, and multiple modes. This is implemented by the tapped delay line.

The fade rate is also controlled by the model. It is modeled by controlling the tap weights with narrow band Gaussian noise whose bandwidth governs the fade rate. As the fade rates depend critically on path geometry, frequency, and the latitude and longitude, then the low pass filter bandwidth is varied to accommodate different fade rates, as desired by the user. A general range of fading has been estimated to be from .05-15 hz (24).

Long term path loss due to absorption and other effects which make up the median long term path loss are not included in the channel model. These are considered variables to be attached to the signal model used with the channel model and specified by the user. The model is not intended to simulate long term propagation loss. These are in the realm of programs such as the HF Propagation Prediction Computer Program developed at the Institute of Telecommunication Science (ITS) in Boulder, Colorado (12), and CCIR Report 252-2 (8).

Delay

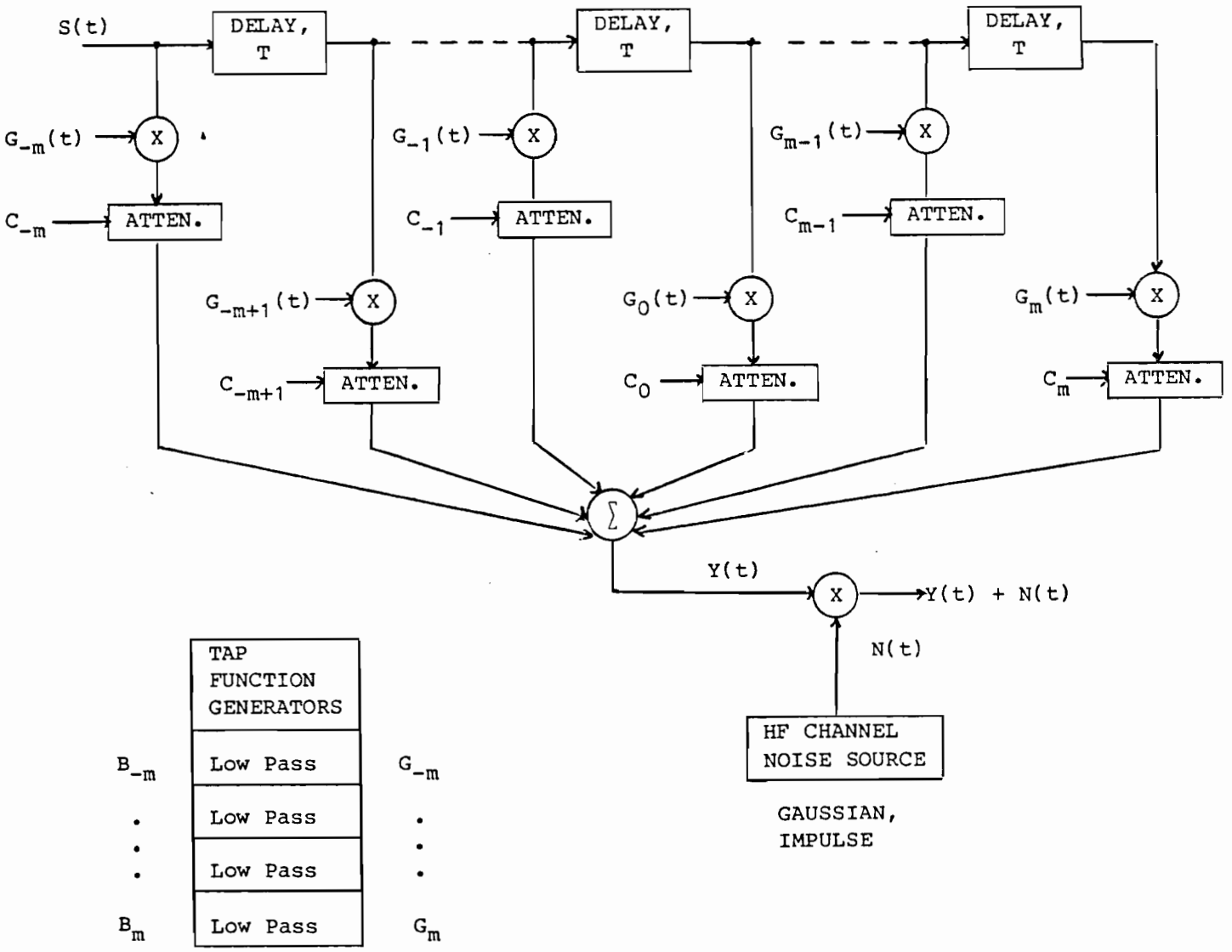
Besides short term and long term fading the other major path phenomena in the model is the delay characteristics of the channel. The model simulates the delay spread within a mode, between modes and for multiple modes. The delay spread has a wide range depending on path and frequency, however is generally in the range of .1 msec. to 10 msec. (19). This is implemented in the model using individual taps and the tap gain multipliers (13, 14).

1.2 HF Channel Model

An HF channel model has been implemented for use in our study of algorithms for an adaptive HF antenna array. The channel model is included in the HF system shown in figure 1.0-1. The channel model uses a tapped delay line and is representative of the HF path as described in section 1.1 of this report and references (1-30). The model we are using is the model developed by Signatron (13) and described in ICS (figure 6, ref. 14). Our implementation of this tapped delay line model is a simplified version as not all characteristics and features of the channel model are necessary for our algorithm investigation. Following sections show a block diagram of the tapped delay line, description of the elements, parameters, software, and example of a simulation.

1.2.1 Channel Model Description

The tapped delay line model used in our algorithm investigation is shown in figure 1.2.1-1. This is the tapped delay line developed by Waterson (29). The elements of the delay line and paramters are the same as the Signatron model (13) and ICS model (14). The model is used in the following manner. An input signal, S(t), is specified, along with various parameters of the channel model which determine the characteristics of the phase interference fading. The outputs of the individual taps are summed and then summed again with a noise source. The noise source is representative of the noise found in the 3-30 MHz band. The resulting output Y(t) + N(t) is a fading, dispersive, noisy signal which simulates a signal received over an HF channel, prior to input to an antenna. The effect of a jammer (undesired signal) is simulated by specifying multiple input signals and processing each signal



S(t) : Input Signal
Gm(t): Tap gain function
$C_m(t)$: Attenuation control (parameter)
N(t) : Channel noise
B_m : Loss pass filter bandwidth (parameter)
X : Complex multiply
Y(t) : Output Signal
T : Tap delay (parameter)

Figure 1.2.1-1 HF Channel Model

through the channel in turn. The user arbitrarily nominates one signal as the desired signal and other signals as jammers. They are then summed in the antenna array as would be the case for an actual system. The user is able to independently specify both the desired Signal to Noise (S/N) and the Signal to Jammer (S/J) ratios. Point of reference for the S/N and S/J ratios are at the output of the channel just prior to input to the antenna array.

Our implementation and specific use is as follows:

Computer System: Model is written in FORTRAN 77. It is run on a Digital Equipment Corporation (DEC) minicomputer, VAX 11/750, with the VMS 4.1 operating system. Our system has an array processor but it is not required to run the channel model. The elements of the model shown in figure 1.2.1-1 are defined as follows:

1. Input signal: The input signal S(t), is not an integral part of the model, but the rate and bandwidth W, of the input signal are used to set certain parameters of the channel model. In general the input signal can be virtually any low pass, deterministic or random data dignal, tone, or noise signal. For our investigation and to test effects of the channel we used a random binary, NRZ, bit stream (low pass equivalent of a BPSK waveform).

2. Delay (T): T is the tap delay of an individual tap in the tapped delay line. The smallest resolvable delay T, is set at 1/2W, where 2W is the Nyquist rate for the transmitted signal of bandwidth W. As the output need only be sampled at a rate of 2W, then a delay of 1/2W is suitable for modeling.

The series of delay elements represents the modes of transmission encountered in the HF path and the time delays associated with each mode. The time delays between modes can be set to an integer multiple of a tap delay, T. For dalays between modes greater than T, multiple tap elements may be included between desired taps. The outputs of the

intermediate tap elements (between desired taps) are not collected and summed. The result is that only the time delay of the signal is affected as it propagates down the tapped delay line. The time delay between modes can be set at some NT, where N is an integer in the range of 1-1000.

3. Multiplier: Provides a complex multiply for each individual tap delay output. The controlled input from the tap function generator is a complex, zero mean, Gaussian tap gain Gm.

4. Attenuator: Provides level control for the signal from each mode (desired tap box in the line of taps) and controls the relative contribution of that mode to the output. This can be set to zero as a means of eliminating the outputs form tap boxes other than the ones desired (intermediate boxes as explained in section on delay). When choosing the attenuator levels for the desired modes, one is in effect setting the delay profile for the channel (14).

5. Summer 1: A linear, infinite bandwidth summing device to add the signals from each of the desired tap boxes in the tapped delay line.

6. Summer 2: A linear, infinite bandwidth summing device to sum the output of the channel with the channel noise source (Gaussian and impulsive noise generator).

7. Tap Function Gen: Additive White Gaussian Noise Generator (AWGN) which serves as the source for the tap gain functions, Gm. For our implementation this is the usual random number generator.

8. Low Pass Filter: Low pass filter between the output of the AWGN and the individual tap gain multipliers. The effect of the filter is to limit the rate and degree of fading which is simulated by the Gaussian tap gain functions, Gm.

9. Noise Generator: A module to add noise to the signal at the output of the channel model. As the noise in the HF band which is received along with the signal is unique for this band (3), then the noise generator module is included as part of the channel model. The noise generator for our initial implementation is Gaussian. However, we plan to add an impulsive noise generator for the final version of the model.

1.2.2 Channel Model Parameters

The channel model in figure 1.2.1-1, is simulated by a software system. The model is operated using an input data stream, specifying channel parameters and collecting the output data stream. As the input and output data stream are an arbitrary string of signaling elements (such as +/-1, 0/1) then the main problem in operating the model is specifying the parameters of the channel. These parameters are (usually) specified in accordance with the actual characteristics of the HF channel one wishes to simulate. The characteristics which can be simulated are generally as shown in section 1.1 of this report. Many characteristics of an HF channel may be simulated. However, for purposes of adaptive algorithm evaluation and algorithm simulation the characteristics and parameters which are important are shown below. These are related to the model in figure 1.2.1-1 as follows:

Channel Characteristics	Channel Model Functions
Fade rate	Low pass filter bandwidth
Propagation modes	Number of taps
Mode attenuation	Tap attenuator
Intermodal delay	Delay T
Delay spread	Tap function generation

An HF channel is simulated by specification of the channel parameters as outlined above. However, an HF channel's parameters vary considerably depending on the distance, time of year, frequency and other path dependent variables. This causes a wide range of values that one could use in simulation.

For the case where one does not have a particular HF channel for modeling then it is useful to have a typical set of parameters one could use. The alternative would be to run through the whole range of parameters which would be costly. For general interest (for a nonspecific channel) we have included some typical values for HF channels. These values have been taken from our HF experience and from the references listed at the end of this report. In particular: Dolukhanov, 3; Davies, 9; Folkstead, 10; Brayer, 5; Goldberg, 19; Ehrman, 22; Waterson, 29; Signatron, 13; and CCIR, 7.

For the specific case (for a defined channel), and also the general case (for an undefined channel), one must specify some parameters to run the model. For both of these cases we have developed a list of the parameters to be specified and the definition for each of the parameters. This is given below in parameter definitions and HF parameters of general interest.

Definition of Parameters (reference figure 1.2.1-1)

1. Signal Bandwidth, W: This bandwidth is normally taken as 4X the bit rate. (Signal bandwidth = 2x bit rate; Nyquist bandwidth = 2x signal bandwidth.) This bandwidth parameter is used (internal to the program) to set the minimum tap delay T. The minimum tap delay T is set at 1/2W. This is the smallest increment of tap delay which is set for simulation. The user specifies W, in KHz.

2. Modes (no symbol): The number of modes refers to the number of major resolveable transmission paths (such as 1E, 2F2, 1F2, mixed). Past experience and channel measurement programs (22, 29, 31) show that no more than 3 modes are necessary to simulate a typical sky wave channel. The user specifies the number of modes as an integer N (typically 3).

3. Sampling Rate (no symbol): The input data is specified by signaling elements per second or bits per second. Sampling rate is specified in samples per second of the input data, S(t). The lowest allowable sampling rate is the Nyquist rate of the input signal which is 2X the input signal bandwidth (W). The sampling rate typically used in simulation studies however is 10X-20X the Nyquist rate (2W). The sampling rate is

used to control the time delay between taps by the ratio I as explained below. The Nyquist rate is used to set the minimum tap delay T, in each tap element. The ratio I is the ratio between the sampling rate and the Nyquist rate. This ratio sets the minimum time delay between taps. When the sampling rate is minimum (equal to the Nyquist rate), I is set to one, and the minimum time delay between taps is 1/Nyquist rate. For integer values of I the time delay between taps is increased as integer multiples of 1/Nyquist rate.

I = Sampling rate/Nyquist rate

I = integer value of the ratio with:
minimum value = 1 and maximum value = 100

I as defined is normalized to the Nyquist rate. For I = 1 (minimum value) the input signal is taken from adjacent tap elements each of which is set at the minimum delay interval T. For I > 1 (for example 4), the input signal is taken from every 4th tap element in the tapped delay line, giving an intermodal delay of 4T. Thus given the Nyquist bandwidth (2W) of the input signal S(t), and the sample rate the program calculates:

T = 1/2W and I = Sample rate/2W

T = element tap delay

W = Nyquist bandwidth (rate)

Note: The sample rate is also used to normalize the 3 dB bandwidth of the low pass filter in the tap function generator. The user specifies the sampling rate in KHz.

4. Fade Rate (Bm): The tap gain functions, G_m, are output from the individual low pass filters with bandwidth Bm. The low pass filter limits the rate of change of the Gaussian random variables used to derive the Gm tap gain functions. The bandwidth Bm, sets an upper limit on the multipath fade rate. The bandwidth is normalized to the sampling rate as follows:

Normalized Bm = Bm (actual)/Sampling rate

Bm ranges from a minimum of 0 implying no fading. To a maximum of .5 implying fast fading at the rate of generation of the tap functions G_m. The maximum rate G_m coefficients can change is half the sampling rate which is equal to the input signal bandwidth. The user specifies a normalized value between 0.0 and 1.0.

5. Attenuator Control (Cm): The output of each tap box goes through an attenuator controlled by the Cm coefficients. The range of Cm is from 0.0 representing infinite attenuation to 1.0 representing no attenuation. These attenuators are used when it is desired to have a different median path loss for one mode in relation to other modes. If it is desired to apply equal path loss to all the modes, then all attenuator coefficients would be equal or 1. In the extreme case, an attenuator coefficient could be set to 0.0 to completely eliminate the signal from one mode. The user specifies the value between 0.0 and 1.0.

6. Tap Gain Functions (Gm): These are complex, zero mean, wide sense stationary, Gaussian processes. The rate of change of these functions is set by Bm, the low pass filter bandwidth. The Gm are generated internally by the program and no definition is required from the user. The Gm are updated at the sampling rate. This rate provides new values of Gm each time a new sample of the input is passed through the tapped delay line.

HF Channel Parameters of General Interest

There is a wide range and many combinations of parameters that can be used for HF channel simulation. However unless a specific channel will be simulated, it is convenient to classify or partition HF channels into general categories so that one does not have to deal with the entire universe of parameters. This provides a convenient method of simulating performance of an HF channel without predicting actual channel parameters or without running the simulation for all possible parameters. This method is recommended by the

CCIR (International Radio Consultive Committee) in one of their major publications, Fixed Services at Frequencies Below About 30 MHz, Volume III, Geneva 1982. This is described in:

Report 203-1, Path Time Delays and Shifts Caused by Multipath Propagation on Radio Circuits

Recommendation 520-1, Use of High Frequency Ionospheric Channel Simulators

This report and recommendation is based upon many observations and data from telecommunication administrations and others all over the world who have an interest in HF systems. It has been discussed during many meetings of CCIR study groups and is a consensus of a reasonable way to classify channels and their parameters. There have been other reports (3, 5, 9, 10, 13, 19, 22, 29) citing many different channel parameters based on channel observations and the relationship to the channel model. However CCIR Report 203-1 and CCIR Recommendation 520-1 are the only publications, with wide distribution, which provide general categories for channels and parameters. Therefore we find it appropriate to list the CCIR channel parameters and categories as a guide for HF channel simulation when one is interested in general effects and general performance of the channel model. These categories could also be used to classify a channel if one knows apriori the characteristics of a channel. The CCIR categories are given in terms of the dealy profile as this has the biggest single effect on channel quality, all other variables being equal. CCIR Recommendation 520-1 uses the following:

General Category of Channel	Parameters	Mean
Good	Delta T	.5 msec
	Intermodal atten.	0dB
	Fade rate	.1 Hz
Moderate	Delta T	1 msec
	Intermodal atten.	0 dB
	Fade rate	.5 Hz
Poor	Delta T	2 msec
	Intermodal atten.	0 dB
	Fade rate	10 Hz
Flutter	Delta T	.5 msec
	Intermodal atten.	0 dB
	Fade rate	10 Hz

Note:

1. Delta T refers to mean time delay between modes of propagation.

2. Intermodal attenuation refers to the difference in median path loss between different modes of propagation.

3. Good, moderate, and poor refer to the relative quality of the channel and is primarily a function of the delay. Flutter is a channel chracterized by excessive Doppler due to rapid changes in scatterers in the path.

4. Common characteristics (which can remain the same for all channels or varied by the user depending on particular requirements):

   - Gaussian noise
   - Multipath
   - Power fading
   - Intermodal attenuation could vary from 0 to -40 dB, but for general categories of channels, path loss for all modes is generally assumed to

be worst case which is 0 dB.

1.3 HF Channel Model Software

The channel model software is built around the tapped delay line, random signal generators, input and output modules, and user interface modules. The software routines are integrated and controlled under one main line routine called HFCONTROL. The software is written so the channel simulation can be run either by itself or as part of a larger program containing other parts of a system such as antenna, receiver and adaptive algorithm. The routines are shown in figure 1.3-1, Channel model Software Routines. A description of the routines, below, in conjunction with figure 1.3-1, provides some insight into use of the program. Figure 1.3-1 shows the hierarchy of the routines while the definitions explain the routines and their usage.

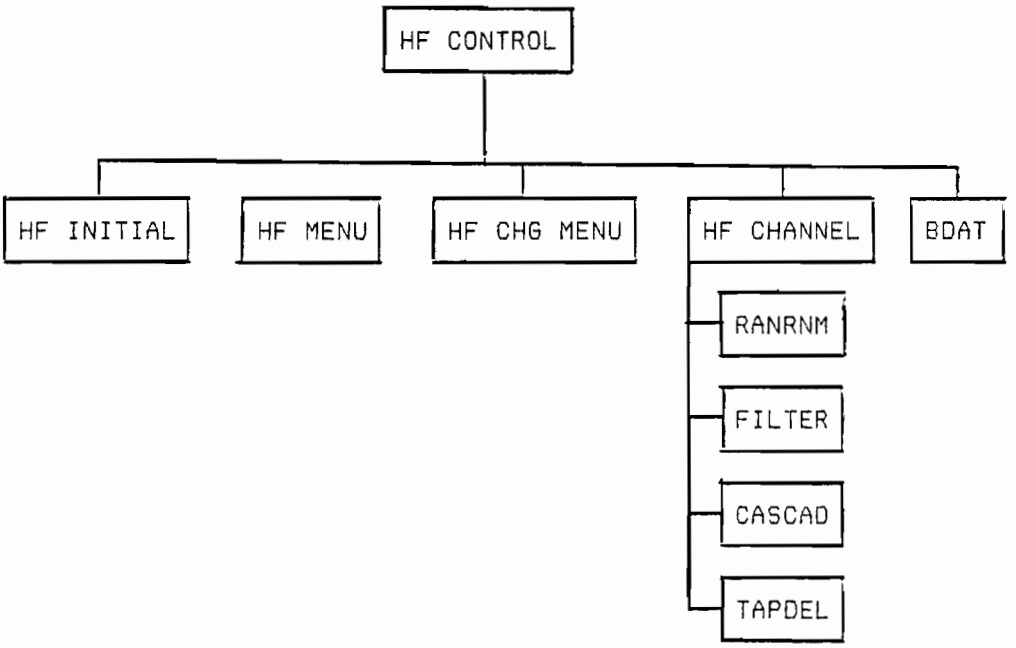


Figure 1.3-1 Channel Model Software Routines

Software Routines Definitions

HFCONTROL: Main line routine for control of the channel simulation. HFCONTROL allows the user to enter channel parameters or use default values, loads the initial channel parameters specified by user, runs the simulation with the channel model, collects the output data, and allows data to be saved and plotted and/or printed. HFCONTROL does this by calling other routines as well as controlling the overall simulation. Other routines called from HFCONTROL are: HFMENU, HFCHGMENU, HFINITIAL, HFCHANNEL, and BDAT. These are explained below.

HFMENU: This routine is used to collect and display the channel parameters used for simulation. It has the default values which can be used or changed.

HFCHGMENU: Used to run the simulation with values other than the default values. It is a menu driven routine that prompts the user for his parameters.

HFINITIAL: Transfers the channel parameters from the HF menu to the channel model and initializes the model with the user's parameters.

HFCHANNEL: The routine that simulates an HF sky wave channel. It uses the parameters loaded by the initialization routine and input data stream as specified by the user. This routine consists of several subroutines for:

Random Numbers: for the tap functions, Gm

Filter: for the low pass filter in the tap function generator.

Tapped Delay Line: for the tapped delay line

Summing: to sum the signals as they are output from the tapped delay line.

BDAT: Generates a file to save the output from the channel model during simulation. The file data may be used for plotting the output data stream from the channel.

The use of these routines is shown in figure 1.3-2, Channel Model Flow Chart. This flow chart shows the flow of control and order of processing of the program. The implementation is as shown following the flow chart diagram.

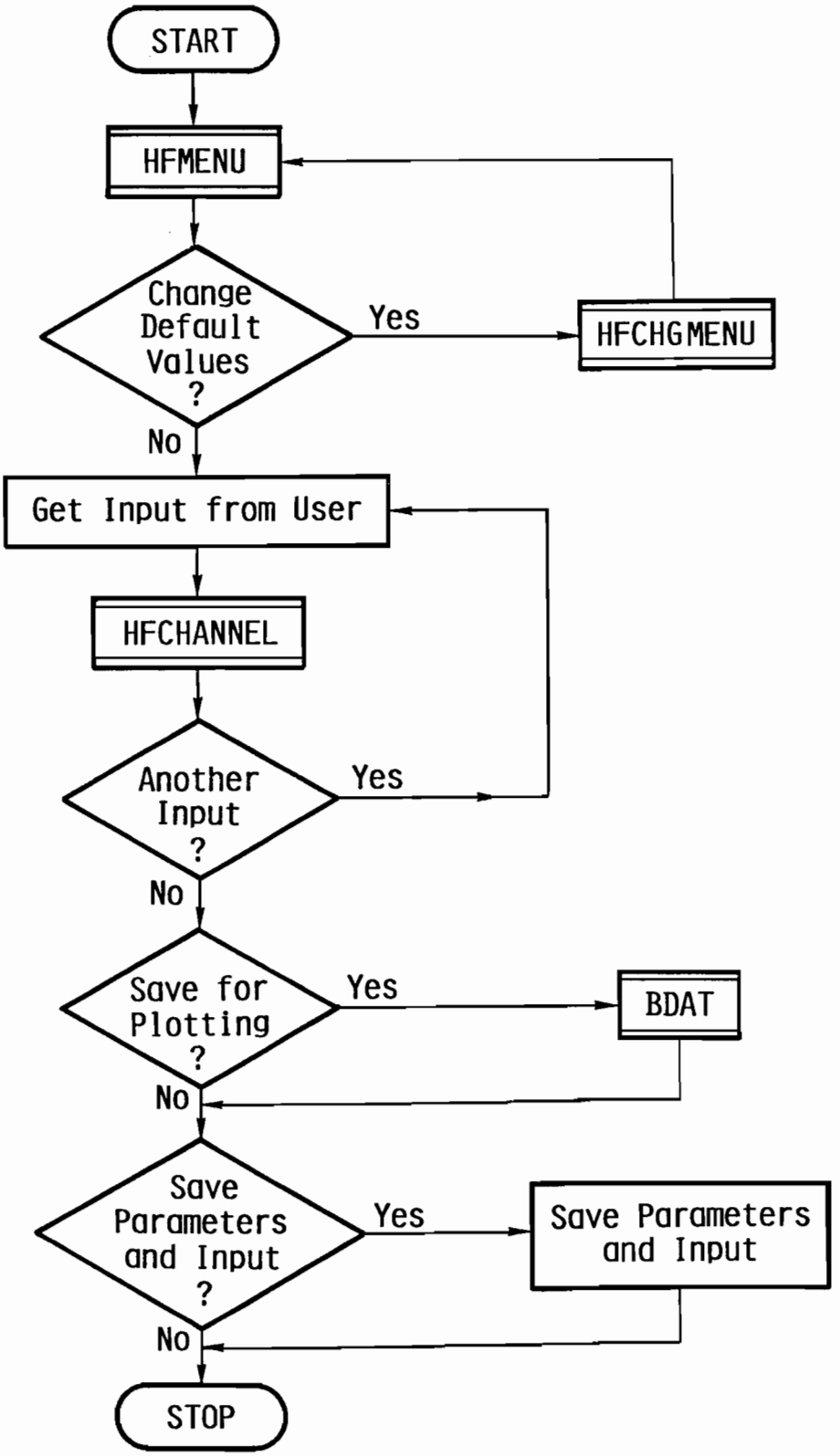


Figure 1.3-2 Channel Model Flow Chart

Program Processing

Step	Action	Subroutine
1.	Start up.	HFCONTROL
2.	Display menu.	HFMENU
3.	Use default values or change to new values	HFCHGMENU
4.	Initialize the model with parameters from step 3 and display values being used.	HFINITIAL and HFMENU
5.	User specifies his input data stream.	HFCONTROL
6.	Input data stream from step 5 is run through the channel model using channel parameters displayed in step 4.	HFCHANNEL
7.	User can stop or supply another input data stream or another set of channel parameters.	HFCONTROL
8.	User collects output data stream for plotting or can discard.	BDAT
9.	User can save channel parameter set used in simulation and can print or discard.	HFCONTROL
10.	End and return to main menu.	HFCONTROL

As the user steps through the program under HFCONTROL, either HFCONTROL or one of the other routines provides prompts and displays values and status of the simulation as it proceeds. A summary of the simulation is as follows:

- Select and specify channel parameters.

- Input user's data stream to the channel.

- Run channel simulation with desired parameters.

- Collect output data stream from the channel simulation.

- Plot output or discard and print the channel parameters or discard.

1.4 Channel Simulation and Results

A system simulation using the channel model in figure 1.2.1-1 was run. the channel parameters used were:

1. Signal Bandwidth, W: 600 HZ

2. Modes (no symbol): 3

3. Sample Rate (no symbol): 6000 HZ

4. Fade Rate, Bm: 1 Hz

5. Attenuator Control, Cn: Varied during simulation

6. Tap Gain Functions, Gm: Generated internally at the sampling rate.

A plot of the input data superimposed on the output data stream is shown in the following figures 1.4-1 through 1.4-4. These figures show the effect of changing the magnitude of the attenuator functions Cn. There are other effects which could be shown such as changing the fading rate (Bm), sample rate, input signal bandwidth (W), or any combination of these items.

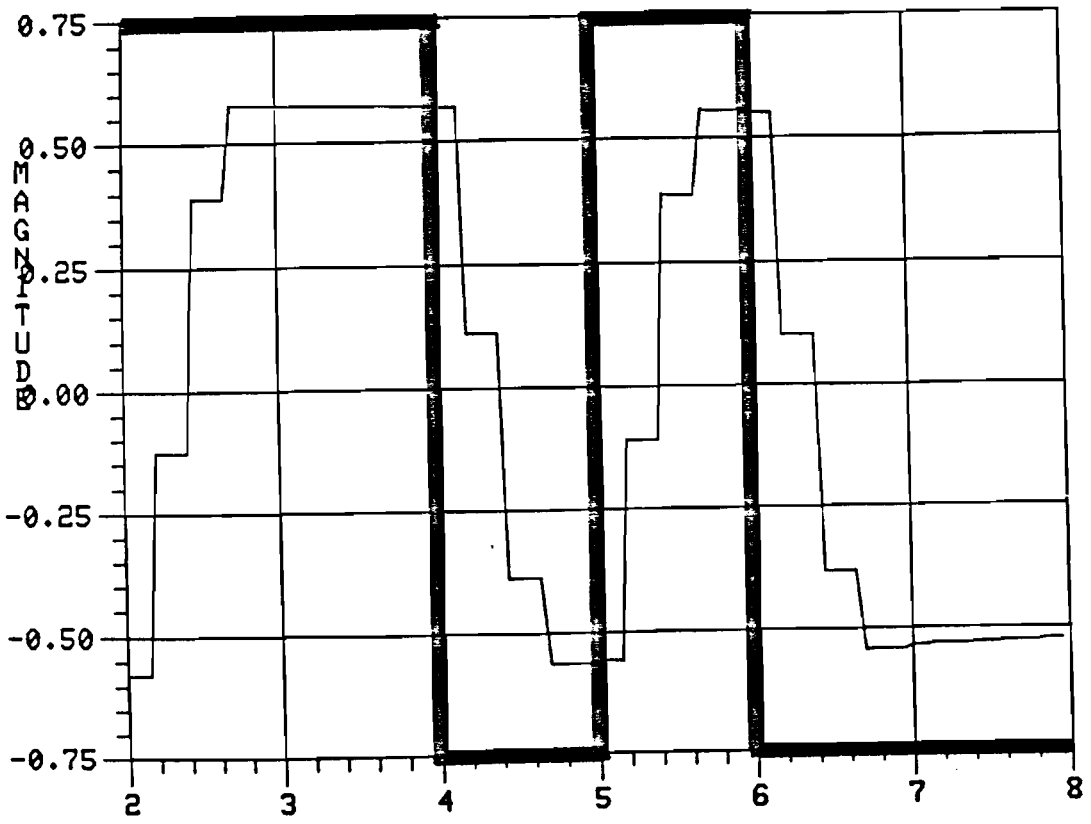


Figure 1.4-1 Channel Simulation Equal Tap Gains

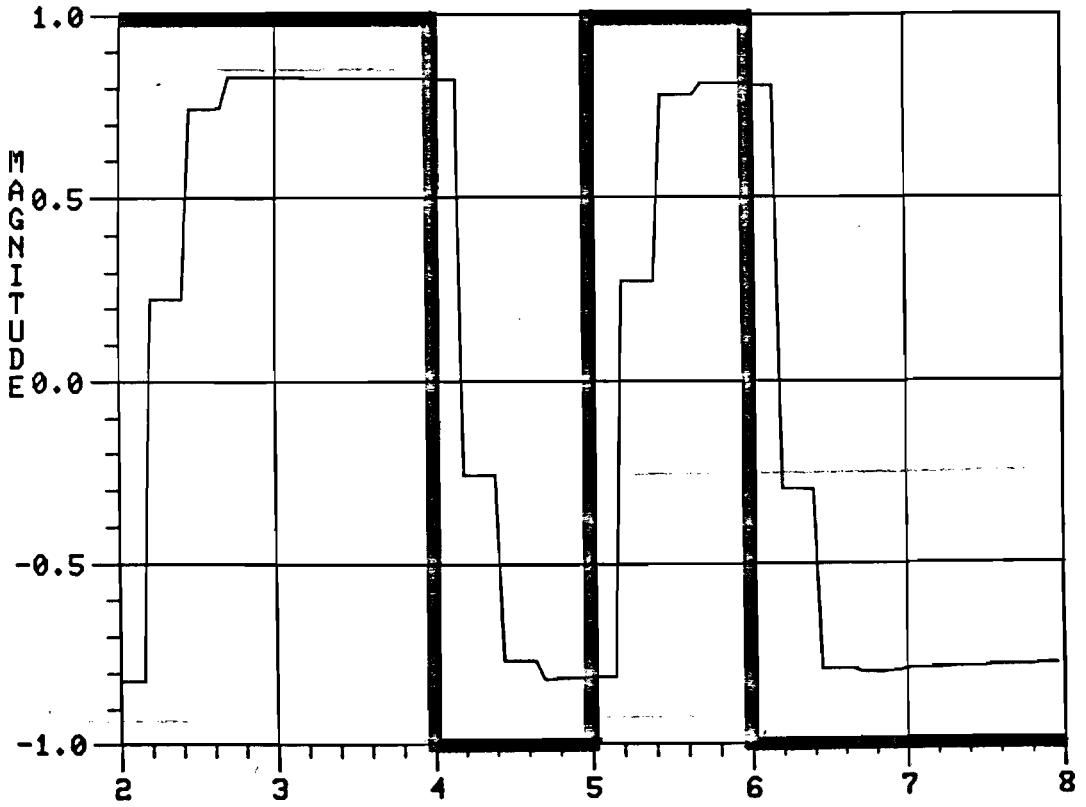


Figure 1.4-2 Channel Simulation Tap 1 Attenuation 2X Tap 2 and 3

Figure 1.4-3 Channel Simulation Tap 2 Attenuation 2X Tap 1 and 3

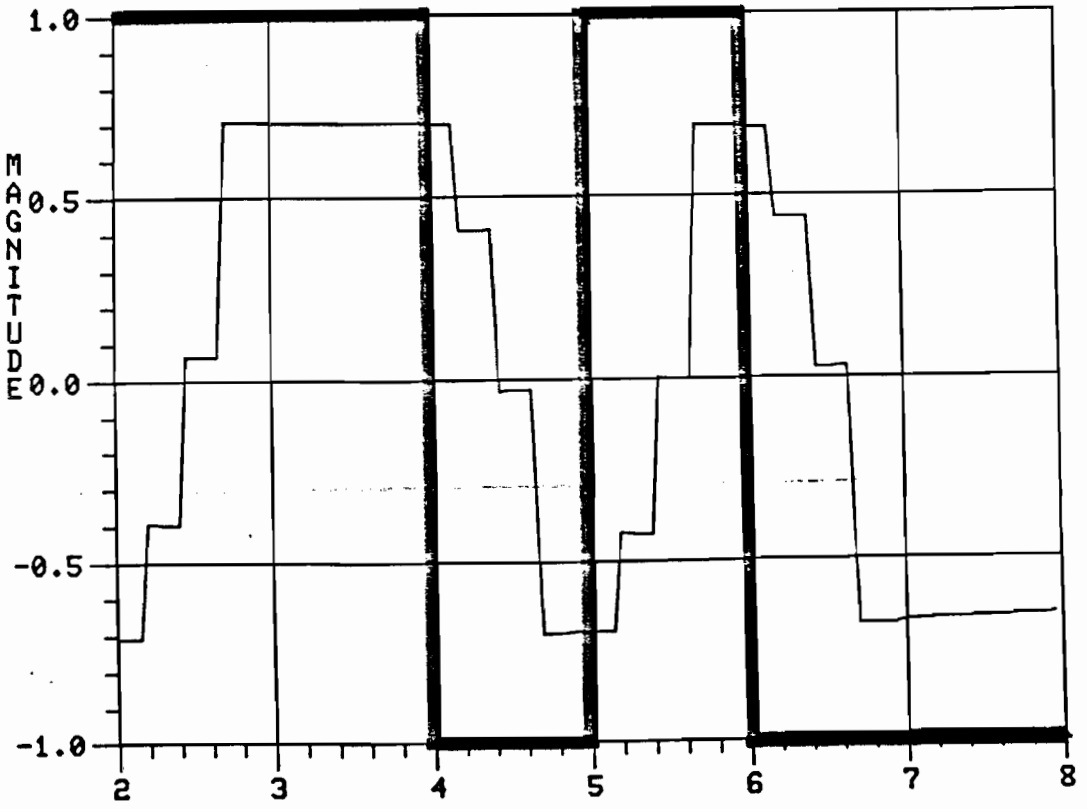


Figure 1.4-4 Channel Simulation Tap 3 Attenuation 2X Tap 1 and 2

1.5 Review and Summary of Task 1

We have developed, defined and characterized various aspects of the HF channel. We concentrated on the HF channel in the beginning because the channel is unique and must be fully understood and considered when choosing and implementing an adaptive algorithm for an HF antenna array. The HF channel details have been developed on the basis of published material and known quantities describing the channel. We have not developed anything new but only applied known details that are consistent with successful HF systems.

The work done in Task 1 generally falls into three areas:

1.1 Literature review, summary and characterization of parameters and models.

1.2 Application of parameters and models to a specific model for this program.

1.3 Implementation of model and simulation of channel with data transmission.

The results of Task 1 are an HF channel model which can be used in Tasks 2 - 5 for investigating and developing adaptive algorithms. The main use planned for the model is to:

- Use as a source of channel data to test and check candidate algorithms.
- Use in a system simulation to test, check and validate various algorithms.
- Serve as a demonstration system to run algorithms and be able to observe the algorithm performance.

The chief purpose of the channel model is to help ensure the candidate algorithms are suitable without requiring use of an actual HF channel for testing. This is accomplished by using the model as a source of channel data and in system simulation.

Task 2

Review of Arrays and Algorithms

Task 2.1 Arrays

2.1.1 HF Antenna Arrays

2.1.2 HF Antenna model

2.1.3 HF Array Software

2.1.4 Simulation

Task 2.2 Processing Algorithms

2.2.1 Overview of Adaptive Algorithms

2.2.2 Adaptive Algorithm Model and Parameters

2.2.3 Comparison and Analysis of Adaptive Algorithms

2.1.1 HF Antenna Arrays

An HF antenna array is included in the system model used for simulation. The antenna array used in the system is shown in figure 2.1.1-1, Planar Array Model. This is a model of a planar array of dipoles, En. Signals, Sm, are received by the array from any arbitrary azimuth and elevation angles. The antenna array is required in our investigation of an adaptive algorithm for:

- Testing
- Simulation
- Collecting Performance Statistics
- Validation of Algorithm Model
- Evaluation of Algorithm

The array is being controlled by the adaptive algorithm. The changes in the array pattern, the speed of changes, and the relative quality of the combined output signal from the array will give us a measure of performance of the algorithm. Using the simulation models this can be examined under various conditions. The array serves in this way as a test bed for the adaptive algorithm. As we want the algorithm investigation and results to be relevant to other work being performed at RADC, we have used an array similar to the one being built at RADC for another phase of this project. This similarity allows a measure of confidence between the results of our investigation and actual measurements and performance tests to be done at RADC at a later time.

2.1.2 HF Antenna Model

The planar antenna array model in figure 2.1.1-1 has an equivlent electrical model shown in figure 2.1.2-1, Antenna Block Diagram. In the block diagram, the individual signals (H1-Hn) received from the individual elements (E1-En) are weighted (W1-Wn) and summed to produce a single combined output, S(t). The effect of the weighing and summing is to produce some desired pattern with control over: directivity, gain, sidelobe levels, and sidelobe nulls. The element control weights are from the adaptive algorithm under investigation. Both the individual element signals and the combined output are used by the adaptive algorithm to derive these element control weights.

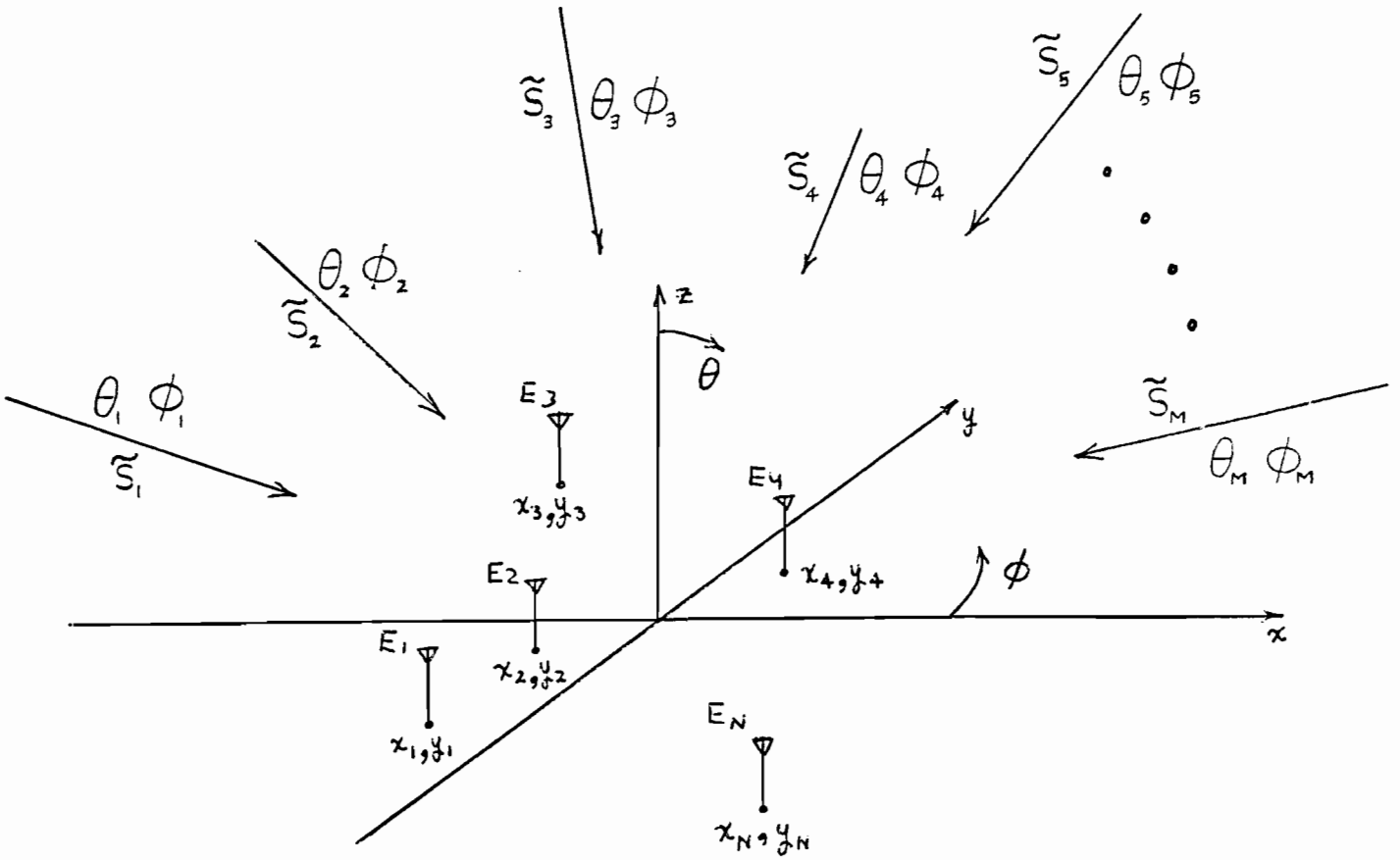


Figure 2.1.1-1 Planar Antenna Array Model

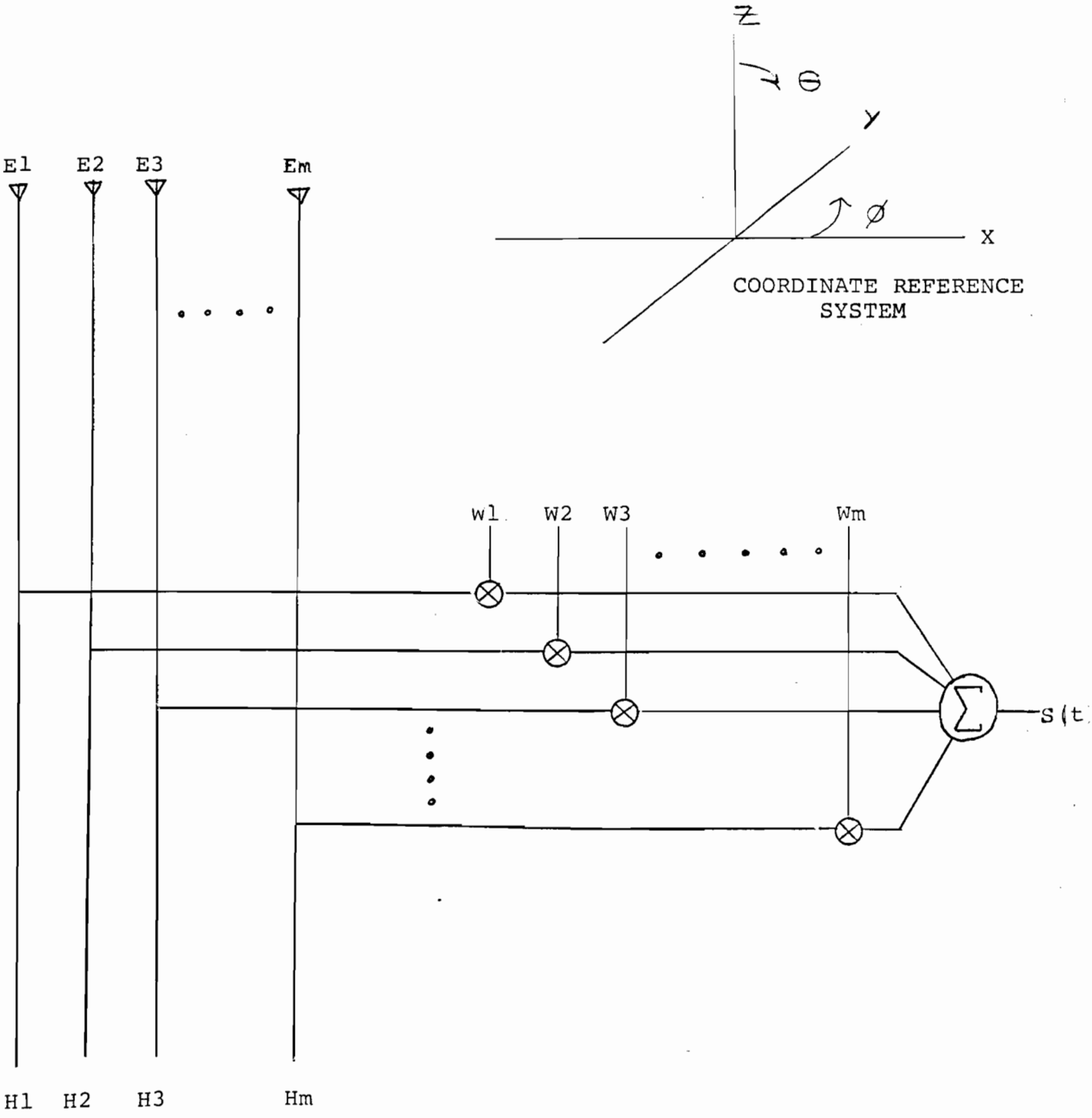


Figure 2.1.2-1 Antenna Block Diagram

Description of the Antenna Model and Block Diagram

The planar array model figure 2.1.1-1 and block diagram figure 2.1.2-1 consist of the following parts:

- Elements: Antenna elements are all identical and equivalant to an end fed whip antenna with a user specified length. The elements are shown in the figure as E1 - En, and at locations Xi,Yi.

- Array: The n elements (E1 - En) form a two dimensional (planar) array, all on a common surface (ground plane). This is surface XY(Z = 0) in the figure.

- Locations: The element locations are specified in terms of the XY coordinates of the element base. Nomenclature in the program is in the form of Xi, Yi, for the ith element.

- Size: Maximum size of the array implemented in our current model is a 3x3 array.

- Coordinate System: The reference coordinate system for all antenna elements and signals is shown inset, in figure 2.1.1-1. This provides a common reference for all angles, distances, phases (delays), and locations.

The antenna array model works similarly to the physical array it represents, except interelement coupling is not included (explained below). Any signal in space is received at each element with phase (delay) a function of the element location. It is assumed that the amplitude of the signal is the same (uniform) on all elements in the array. Each signal is multiplied by a complex weight to adjust the gain and phase (delay) response desired for the model (such as may be determined by an adaptive algorithm). The individual outputs from each multiplier are summed to produce a combined array output. The only difference from an actual array is that interelement coupling (mutual coupling) is considered as zero in the model. This is done to simplify the model and allow us to implement the model and simulate reasonable size arrays

(greater than a 3x3). If mutual coupling was added to the model the model would be very computation intensive and be costly in terms of computer resources for any simulation exercises. And most important is that the investigation of adaptive algorithms is not affected by the presence or absence of mutual coupling in the antenna array. Hence mutual coupling has been excluded from the array model.

Theory of Operation of the Array

The operation of the array is described on the following basis.

- Coordinate System
- Signals in Space
- Individual Element Response
- Complex Multipliers
- Combined Output

1. Coordinate Reference System

The coordinate reference system for both the signals in space and antenna elements is shown in figure 2.1.2-1, Coordinate Reference System. This coordinate system provides for specification of incident signals by azimuth (ϕ) and elevation (θ), and antenna elements Ei by their locations Xi and Yi.

2. Signals in Space

Any signals incident on the array are considered to have originated outside the near field, and are plane waves with a common wave front, and produce the same amplitude signal on all elements. The signals in space are specified by:

- Direction in terms of azimuth (ϕ) and elevation (θ).

- Vector representation of the signal in the low pass equivalent form: A[cos b + j sin b].
- Frequency of each signal.

On this basis, any number of independent signals both friendly and jammer may be specified. They are all added in each antenna element on the basis of superposition. The relative output of each signal however is a function of the frequency, azimuth, and elevation of that signal.

3. Antenna Element Response

Each antenna element is modeled as a constant length, center fed dipole over an infinite ground plane. The response of each dipole is independent of the azimuth of the incident signal. The response is however a function of the elevation and the frequency of the incident signal. Therefore, the response of each element is for both gain (amplitude) G, and phase (delay) α is computed as a function of elevation and frequency of the incident signal.

3.1 Antenna Element Amplitude Response (Gi)

The amplitude or gain corresponding to any signal incident on any element is:

$$G(\theta,f,\ell) = k\left[\frac{\cos(\frac{\pi f \ell}{c}\cos\theta) - \cos(\frac{\pi f \ell}{c})}{\sin(\theta)}\right]$$

θ = elevation of incident signal i, in radians, set by user.

f = frequency of incident signal in Hz, set by user.

ℓ = length of dipole. For our model this is fixed at 3.8 meters which is equivalent to a 6 foot end fed whip antenna over an infinite ground plane.

c = speed of propagation; $3(10^8)$ m/sec.

k = arbitrary constant (set to 1 for our model).

The amplitude Gi of the output of each element Ei, for each incident signal Sm is:

$$Gi = Sm \times G(\theta, f, \ell)$$

3.2 Antenna Element Phase Response (αi)

The phase response (αi) of each element i, must be calculated in relation to all other elements, all other signals, and for a common coordinate system. This relative phase response is a function of azimuth (ϕ) and elevation (θ) of the incoming signal, and the position (Xi and Yi) of each element. To obtain the relative phase response of all elements so that the signals from all elements may be added, the phase response for each element is calculated in relation to the origin of the coordinate system. The relative phase response is calculated as follows:

$$\alpha(f, \phi, \theta) = \exp\left[- \frac{2\pi j f}{c} \acute{Xi} \sin(\theta) \right]$$

α = phase of output with respect to origin, for element Xi. For an element at the origin (Xi = Yi = 0) there is no phase shift and the phase of the output equals the phase of the input.

f = frequency, Hz, of incident signal.

$\acute{Xi} = Yi \sin\phi + Xi \cos\phi$

Yi = y coordinate, meters, of element i.

Xi = x coordinate, meters of element i.

ϕ = azimuth angle, radians.

θ = elevation angle, radians.

Combining the amplitude and the phase response, the output of each element is:

$$Gi = Sm \times G[\theta,f,\ell] \times \alpha\,[f,\theta,\phi,\acute{X}i]$$

Sm = incident signal vector.

$G[\theta,f,\ell]$ = gain with respect to elevation, frequency and element length.

$\alpha[f,\theta,\phi,\acute{X}i]$ = relative phase with respect to origin, azimuth, elevation and element location (for either friendly or jammer).

As all signals may be superimposed on any element at different frequencies and angles, the output of any element may contain:

$$Gi = Gi(f1) + Gi(f2) \text{ ------- } + Gi(fn)$$

Gi = output of the ith element

Gi(fn) = response at frequency fn

4. Complex Multiply:

Each signal Gi is multiplied by a complex weight, Wi, which is generated by the adaptive algorithm. The overall result is to steer the beam, control the sidelobes, and produce nulls at desired locations. The signal produced by the complex multiply is:

$$Hi = Gi \times Wi$$

Hi = output of the ith multiplier for the ith signal

Gi = output from the ith antenna element

Wi = control weight from the adaptive algorithm

5. Summed Output:

The output from the array is the summation of the individual weighted signals from each antenna element:

$$S(t) = \sum_{i=1}^{m} Hi = \sum_{i=1}^{m} Gi \times Wi$$

S(t) = summed output

2.3.1 HF Array Software

The array model in figure 2.1.1-1, is implemented in software as a series of algorithms that simulate this array. The program is written in FORTRAN 77. It is run on a DEC VAX 11/750, with the VMS 4.1 operating system. The software consists of a main line program, MARRAYANR, user prompts, and a routine, ARRAYANT, to perform all calculations. The operation of the model is described in the following section.

Model and Program Operation

The program steps simulate the operation of an actual antenna array. The user begins the program by specifying array parameters and the directions of signals in space to be passed through the array. The program calculates the response of the array to the signals in space depending on the frequency, azimuth and elevation. The output of the array is a signal modified by the array response. This provides an input - output relationship for fixed conditions on the array and a signal fixed in space.

A more interesting and useful application is to simulate the pattern of an array as a function of element weights for a given set of signals. This pattern simulation is the usual application of the array program for adaptive algorithm investigation. In this application the antenna array is initialized and a set of signals are specified. The array is temporarily fixed and the signals are rotated about the array varying both azimuth and elevation. This provides the typical polar gain plot of an antenna. When this method is used with an adaptive algorithm the algorithm controls the element weights and the

array program calculates the array response for each set of weights specified by the algorithm. At the end of simulation the user can read either a polar plot pattern or the element control weights.

An alternative use of the array model is in a manual (nonadaptive) control mode. The user specifies and fixes all array parameters and varies the element control weights according to some predetermined distribution (such as uniform, Chebychev, or sin x/x). The antenna array model could be used in this manner for pattern synthesis in antenna design. It could also be used to generate a series of patterns as a function of other parameters besides control weights.

Program Implementation

The implementation of the array model software is shown in figure 2.1.3-1, antenna Model Flow Chart. The flow chart in this figure shows the flow of the program steps from user prompts through to creating an output file. Beginning at the top of the figure the program operates as follows:

Step	Action	Description
1.	Create Signal Structure	User prompt to specify all information about the signal to be processed through the antenna. This is retained in a file as a signal data structure. Items specified are complex low pass equivalent, frequency and sampling rate.
2.	Open Output File	Subroutine internal to program to create a file for output data. User is prompted to specify name of the output file.
3.	Array Data	Prompt for user to specify data about the antenna array. Items specified are number of elements, coordinates of each element, and initial weighting factors (amplitude and phase) for each element.

4.	Input Signals	Prompt for user to specify data about the signals in space. Items specified are frequency of the signals, number of signals (at some frequency), and the azimuth (ϕ) and elevation (θ) of each signal.
5.	Compute Response	Subroutine ARRAYANT computes the array response to signals and array parameters previously specified.
6.	Save Output	Writes all data to an output file, name specified in step 2 above.
7.	Repeat or Stop	Prompt to user to decide to stop and save data or continue. If continuing, the user must specify whether using old data or new signal and new array data.

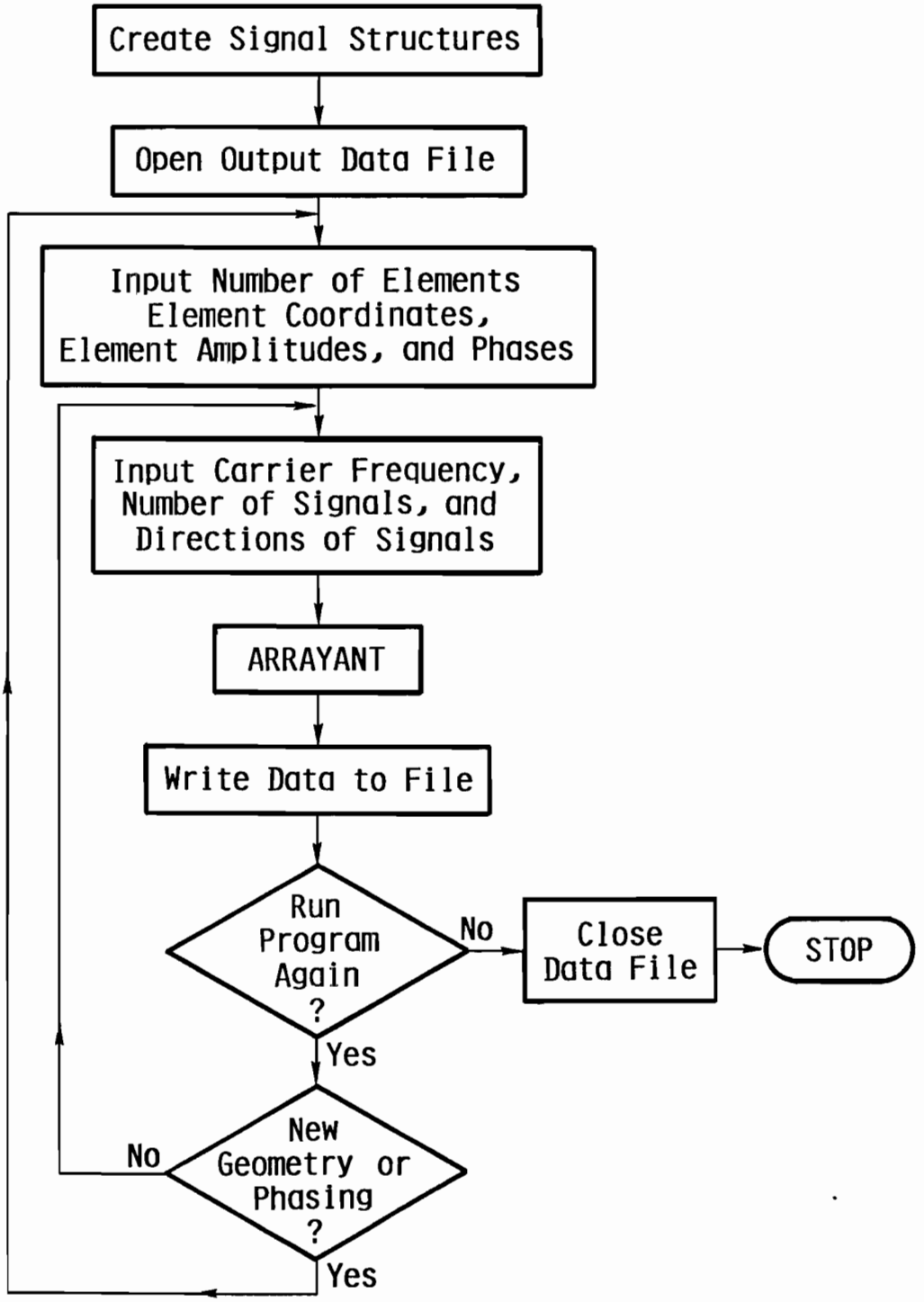


Figure 2.1.3-1 Antenna Model Flow Chart

Description of Variables

The variables and parameters used in the model and specified by the user are shown below. These variables and parameters are as shown in array model diagrams, figures 2.1.1-1 and 2.1.2-1 and the user input menu (on the computer).

Variables	Program Name	Description
N	NANT	Number of dipole elements contained in the array.
M	NSIG	Number of incoming signals.
S1 -- Sm	SIGNAL	Complex low pass equivalent representation of an incoming carrier. It is specified by the user as Acos + Bjsin. It is typed in A, B.
θ	THETA	Elevation angle of the incoming signal in degrees. The convention used is that elevation of 0 degrees is overhead or along the Z axis.
ϕ	PHI	Azimuth angle of the incoming signal in degrees. PHI is assigned a value of 0 degrees on the x axis and increasing in value as the azimuth increases to the y axis in a counterclockwise direction.
F	CFREQ	Carrier frequency of an incoming signal in Hz. For our investigation of adaptive algorithms we assume that all incoming signals have the same frequency.
X1 -- Xn	X	X coordinates of the dipole elements 1 through n.
Y1 -- Yn	Y	Y coordinates of the dipole elements 1 through n.
L	LENGTH	Antenna length, meters.

The output values of the array model and the algorithm are shown below. These are calculated during simulation.

Variables	Program Name	Description
H1 -- Hn	OUTPUT	The complex lowpass equivalent output from each antenna element before multiplication by complex weighting factors.
W1 -- Wn	WEIGHT	Complex weighting factors which are used to adjust the amplitude and phase distribution of the individual antenna element outputs.
S(t)	WTDSUM	Summation of the individual weighted element signals to produce a single combined output.

Summary of the Array Model

An antenna array model is used as part of our investigation of adaptive algorithms. The array is used for testing, simulation, performance statistics, validation of algorithms, and evaluation of algorithms. The array model is similar to the physical array being built by RADC as part of another program. Our model is a planar array of dipoles electrically equivalent to an end fed whip antenna with length specified by user. The array elements are weighted and summed to allow us to control gain, steer the beam, adjust sidelobe levels, and control placement of nulls. The user can simulate an array pattern by specifying array parameters, signal parameters, rotating the signal about the array and plotting the output as a function of azimuth and elevation.

When used with an adaptive algorithm the algorithm controls the weights to produce a desired pattern of the main beam, side lobe and nulls. The pattern obtained, when using an adaptive algorithm, will be a function of the algorithm and the error criteria or measure of goodness built into the algorithm. In addition to array pattern analysis and simulation of various patterns, the array and algorithm is also used to measure the speed of conver-

terns, the array and algorithm is also used to measure the speed of convergence for different algorithms. The speed of convergence is an important performance criteria and is a key factor in our investigation. An example of the use of the array is shown in the following section.

2.1.4 Simulation

The antenna array model was run with an adaptive algorithm in a pattern analysis study to investigate the effect of the algorithm on the array pattern and the resulting improvement in S/J ratio. For this simulation example a jammer was incident on the array at 165 degrees azimuth and 60 degrees elevation. The purpose of the simulation was to adaptively adjust the pattern to minimize the jammer power or maximize the S/J ratio. The initial parameters used for starting the simulation were as follows:

Array: 3x3 rectangular geometry

Elements: 10 meters, end fed whip antennas

Element spacing: 5 meters, uniform spacing between all elements

Frequency: 30 MHz

Amplitude distribution: equal amplitude on all elements

Phase distribution: 0.0 relative phase on all elements

Prior to simulation the antenna array pattern and jammer is shown in the first two figures.

Figure 2.1.4-1: Polar plot of azimuth at a jammer elevation angle of 60 degrees, no adaption from algorithm.

Figure 2.1.4-2: Polar plot of elevation at a jammer azimuth angle of 165 degrees, no adaption from algorithm.

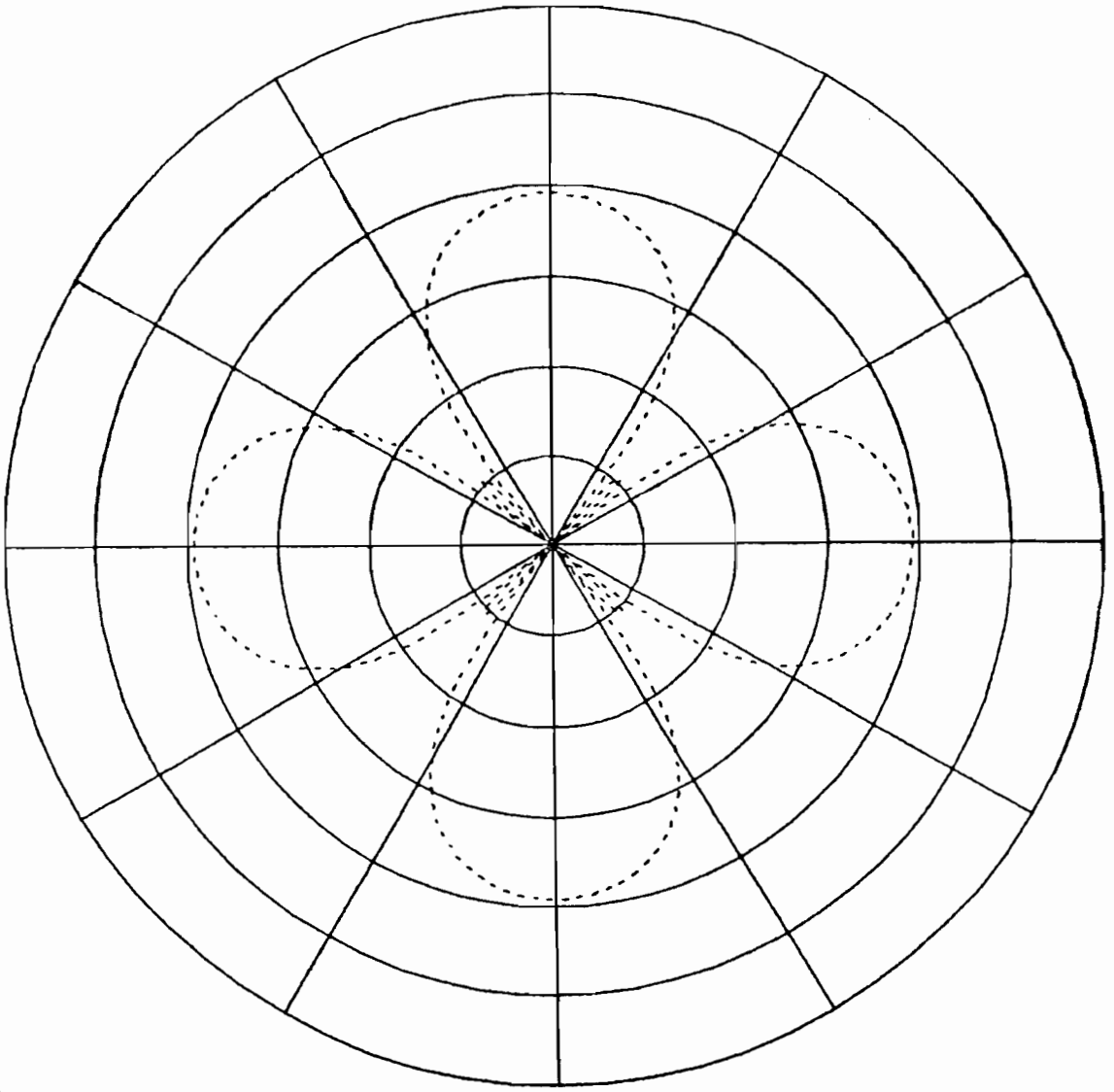


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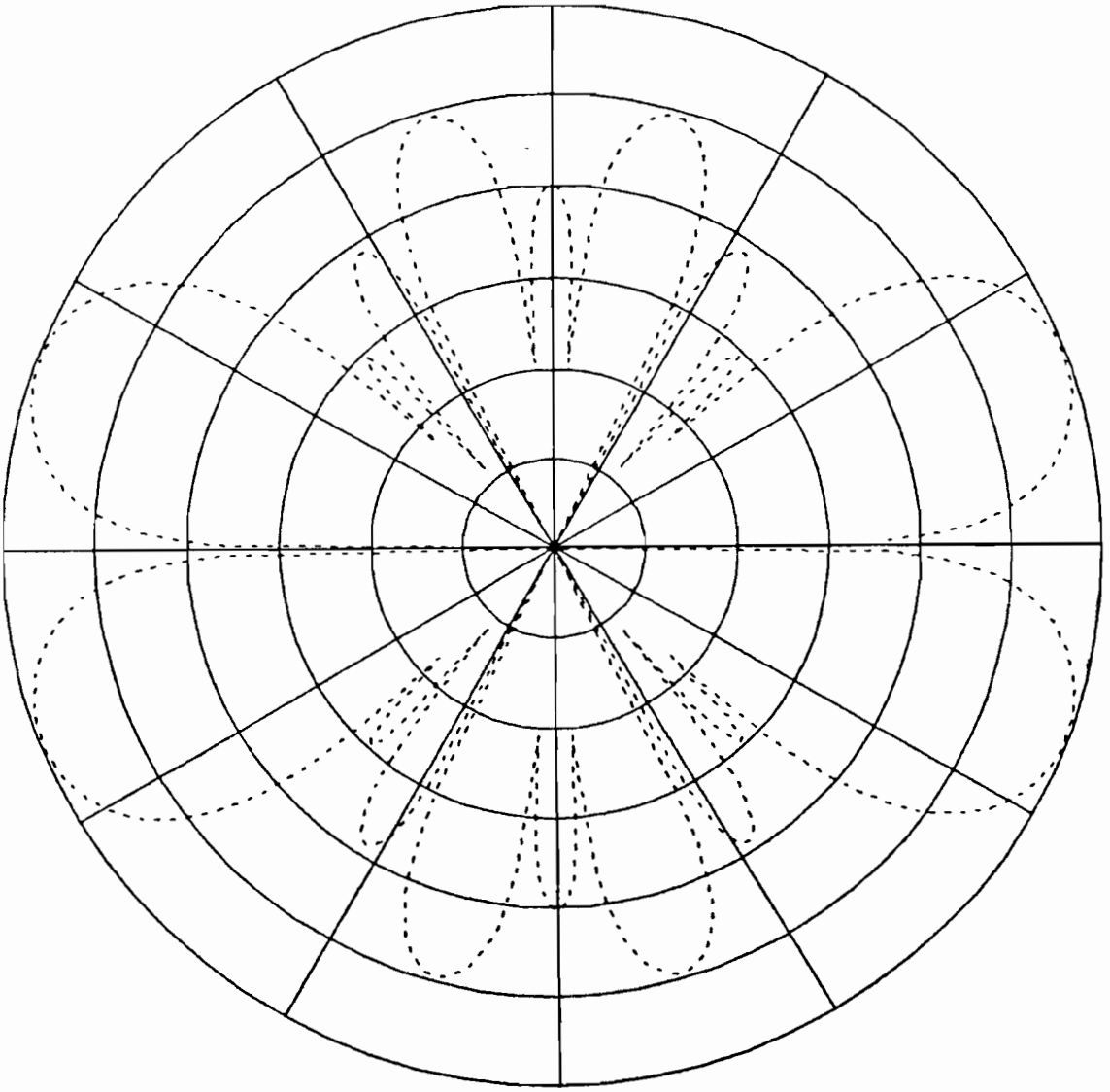
Figure 2.1.4-1 Azimuth Polar Plot, Unadapted, Elevation 60 deg.



FORTRAN STOP

Figure 2.1.4-2 Elevation Polar Plot, Unadapted, Azimuth 165 deg.

The results of the simulation are shown in the second set of figures:

Figure 2.1.4-3: Polar plot azimuth after adaption at the jammer elevation of 60 degrees.

Figure 2.1.4-4: Polar plot of elevation after adaption at the jammer azimuth of 165 degrees.

Results: The first two figures show the antenna pattern before simulation. There is one polar plot showing gain as a function of azimuth at the jammer elevation. A second polar plot showning gain as a function of elevation at the jammer azimuth.

The adaptive algorithm adjusted the antenna pattern to minimize the effects of the jammer or to maximize the S/J ratio. The adapted patterns are the second set of figures. In the polar plot of azimuth there is a null at the jammer azimuth of 165 degrees. In the polar plot of elevation there is a null at the jammer elevation of 60 degrees.

Comparing the patterns before and after simulation the improvement in S/J can be seen by comparing the relative gain in the direction of the jammer both before and after simulation. The overall improvement in S/J is better than 30 dB.

2.1.5 Summary of HF Antenna Array Model

The antenna array model works similarly to the physical array it represents, except interelement coupling is not included (explained below). Any signal in space is received at each element with phase (delay) a function of the element location.

It is assumed that the amplitude of the signal is the same (uniform) on all elements in the array. Each signal is multiplied by a complex weight to adjust the gain and phase (delay) response desired for the model. The individual outputs from each multiplier are summed to produce a combined array output.

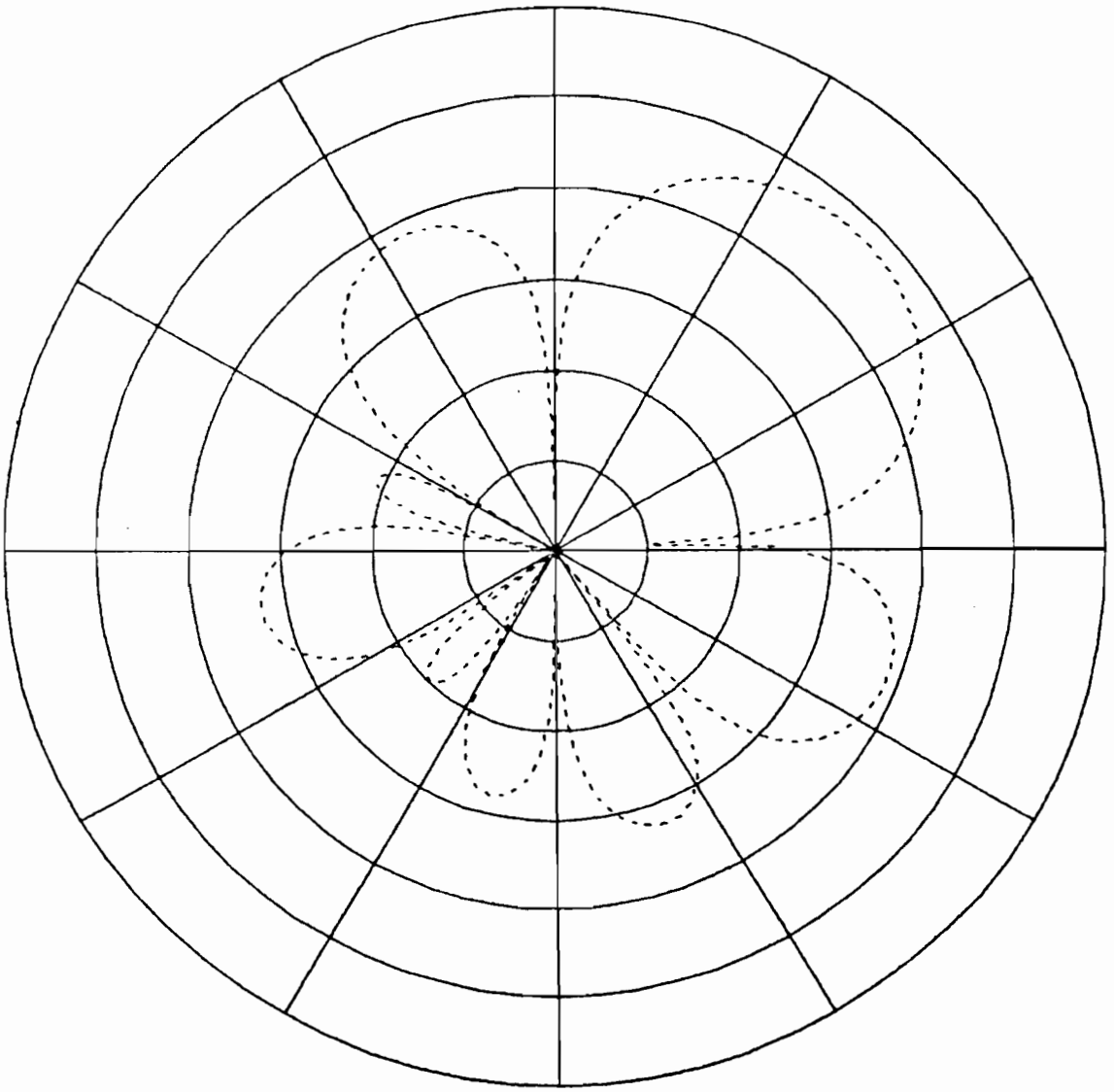


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Figure 2.1.4-3 Azimuth Polar Plot, Adapted Pattern
(Jammer at 165 degree azimuth)

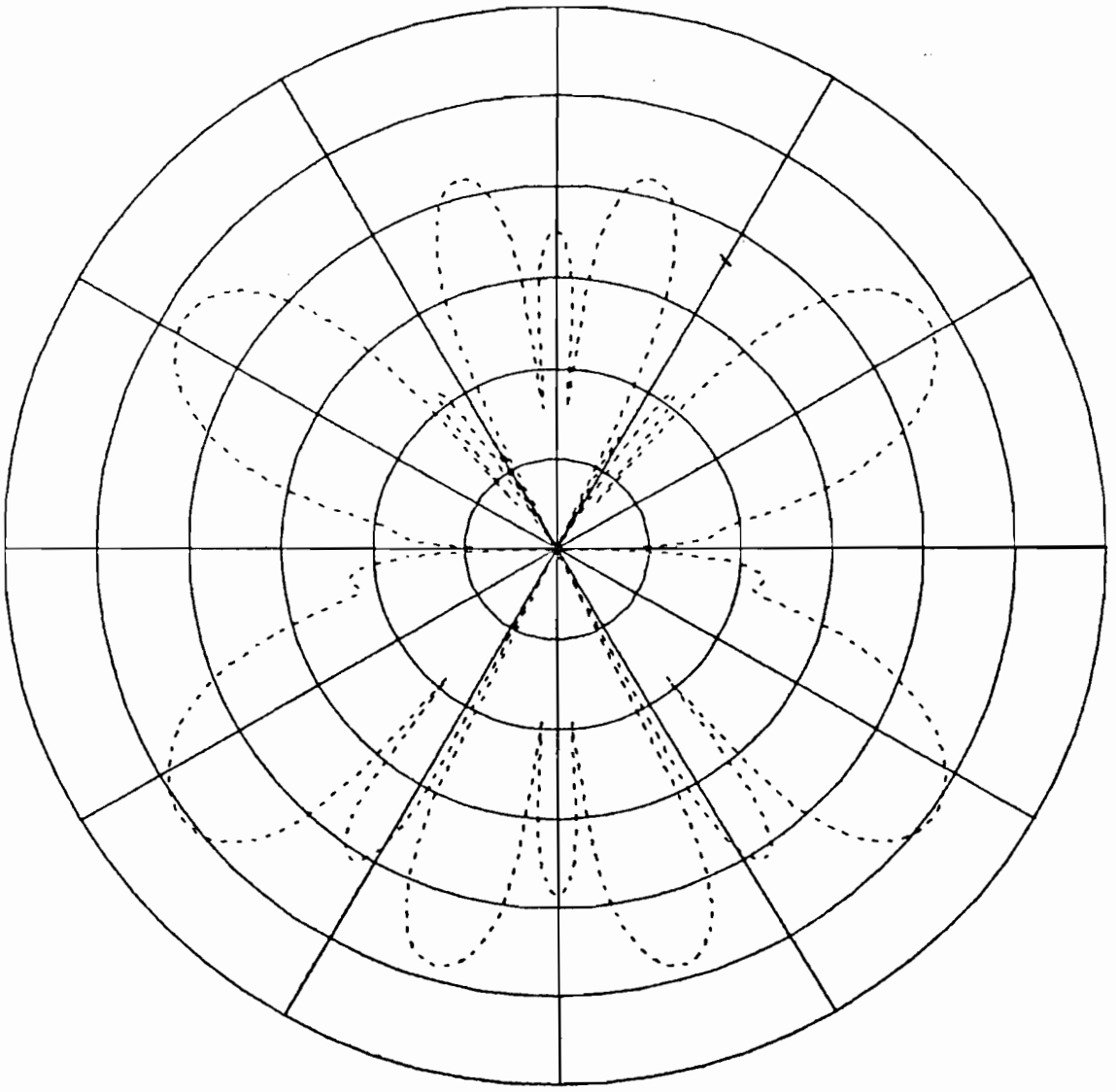

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Figure 2.1.4-4 Elevation Polar Plot, Adapted Pattern
(Jammer at 60 degree elevation)

The only departure from an actual array is that interelement coupling (mutual coupling) is considered as zero in the model. This is done to simplify the model and allow us to implement the model and simulate reasonable size arrays (greater than a 3x3). If mutual coupling was added to the model the model would be computation intensive and costly in terms of computer resources for any design exercises. And most important is that the investigation of adaptive algorithms is not affected by the presence or absence of mutual coupling in the antenna array. Hence, mutual coupling has been excluded from the array model.

2.2.1 Overview of Adaptive Algorithms

The RADC spatially adaptive antenna has the potential for optimizing the received S/N, and minimizing the susceptibility to jammers or other interference by sidelobe reduction and null control. This may be accomplished by appropriate control of the phase and amplitude weights of each of the antenna array elements prior to summing the signal from each element. This control results in steering the main beam to optimize the quality of a desired signal and produce nulls or low sidelobes in the antenna pattern in the direction of jammers or other undesired signals. As the RADC spatially adaptive array is an arbitrary two dimensional array on a plane surface (earth), then one must be able to control the antenna pattern in both azimuth and elevation.

To use the spatially adaptive antenna with a time varying HF channel, and to be able to provide effective, real time, reliable antenna array control, then an adaptive scheme must be used to control the array. Given any reasonable array size operating in the HF environment, and considering that the antenna main beam, sidelobes and nulls must be continuously adjusted for optimum path performance, then only a real time adaptive scheme can achieve any degree of success. The adaptive algorithms being investigated for the array spatial control are specifically evaluated for this environment and use. For the HF channel, they operate in real time, and are not restricted in azimuth nor elevation. The system objective is to maximize gain, minimize interference, and discriminate against multipath. These are the operating criteria for the adaptive processor and its controlling algorithm.

2.2.2 Adaptive Algorithm Model and Parameters

The adaptive systems being investigated are those generally described by S. Applebaum (32), B. Widrow (33), and others (34 - 40). (These are certainly not a complete list of all references on adaptive arrays and processors, but are a useful guide for this application, and include the major ideas and concepts.) A general model of an adaptive system is shown in figure 2.2.2-1, N-Element Adaptive Array Model, based on models defined in References 32 - 40. This is the model used for the investigation of an adaptive processor and adaptive algorithm for the spatially adaptive antenna.

In figure 2.2.2-1, the individual antenna elements are shown as 1 - N, which can be any arbitrary two dimensional array. For the RADC program, and for initial modeling we are using fixed length whip antennas (equivalent to a dipole over a good ground plane). The pattern forming network adjusts the phase from each element and sums the signals to produce the output Y(t). The output of the processor, $W_N(t)$, is a control signal to each element to adaptively control the antenna pattern.

The goal of the adaptive processor as shown in figure 2.2.2-1 is to provide a means to control the antenna element weights Wn(t), in order to optimize the quality of a desired signal, shown here as Y(t). Optimization is with respect to some given quality criteria such as S/N, S/I, or Bit Error Rate (BER).

In general there is no direct nor straight forward relationship between the signal quality in terms of S/N, S/I, or BER and the element control weights Wn(t). Thus the adaptive algorithm must work indirectly to optimize the signal array output Y(t). This is achieved by optimization of some criteria within the adaptive processor (local criteria) which indirectly and apriori, ensures optimization of the array output Y(t). It is the optimization of the local condition (in the adaptive processor) which is described in the following discussion of algorithms, rather than the array output Y(t). The adaptive algorithm is designed in such a way that optimization of the local criteria within the processor, ensures optimization of the array output Y(t).

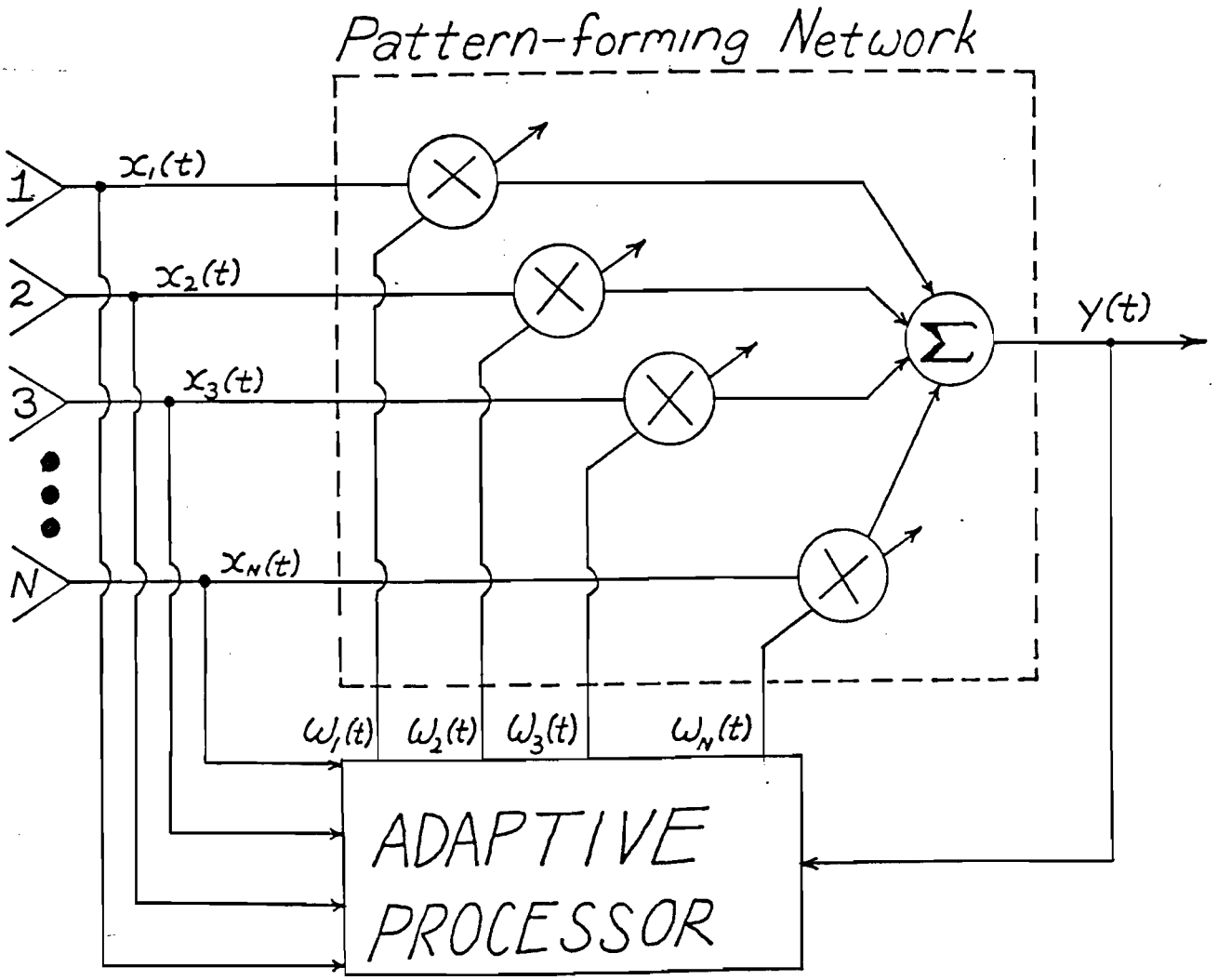


Figure 2.2.2-1 N-Element Adaptive Array Model

The adaptive processor thus serves as a control and processing link between the array output and the array control weights, Wn(t). The results of this operation are to provide, simultaneously: optimum gain for the desired signal; minimum sidelobes to reduce interference and probability of interception; and nulls in the direction of interfering sources. The system therefore depends on the spatial characteristics of the array to achive optimum S/N and S/I. Assuming a well behaved and optimum adaptive algorithm, the limiting factor in the system performance is the spatial characteristics of the array. For a given array, bounds on performance are directly proportional to the maximum achieveable directivity of the array and the maximum achieveable depth of the pattern nulls. Additional bounds on performance are set by the requirement to simultaneously achieve maximum directivity, minimum sidelobes, and maximum depth of nulls. As these are related to array parameters, then the control algorithm must be able to balance these conditions to arrive at the best possible signal output Y(t).

Limits on the spatial array performance also dictate corresponding bounds on the signal environment. The correspondence is between pattern characteristics and separation in space (angle) and time between a desired signal and any and all interferers. If the angle of arrival (or time difference) between two signals is not resolvable in the spatial array, then the array pattern control can not provide any signal separation. There is a similar bound set by the receiver. There must be some minimum separation in frequency and/or signal format between the desired signal and any and all interferers. If the signals are indistinguishable in frequency and/or format then the receiver can not provide any separation either.

2.2.3 Comparison and Analysis of Adaptive Algorithms

Most algorithms fall into two broad categories: iterative and recurive. As used in this report, recursive and iterative refers to two general methods of deriving the optimum weight control vector to satisfy the Weiner-Hopf relation:

$w = R^{-1} P$

w = optimum, element control weight vector

R = autocorrelation of input signal (probe signal)

P = crosscorrelation between input and reference.

The most useful way to define these two methods is in a relative manner rather than an absolute definition of "recursive" or "iterative." What is relevant is the distinction between the interative and recursive algorithms and the method of deriving the optimum weight solution to the Weiner-Hopf relation (objective of each algorithm) rather than a strict definition of each method.

The iterative algorithms derive the element control weights based on an indirect measurement of some criteria (such as the MSE) which is a function of all previous input values (of the probe signal). The method is implemented in an iterative (step by step) manner which asymptotically approaches the optimum solution. The key distinction is that the optimum solution is obtained from an indirect measurement of performance; the optimum solution is not directly calculated.

The recursive algorithms derive the element control weights based on a direct measurement of all previous signal inputs and the algorithm outputs (element weights) to calculate an updated (improved) element control weight vector. This is also performed in a step by step manner. Except in the recursive case the solution to the Weiner-Hopf relation ($W = R^{-1} P$) is calculated each time the element control weight vector (W) is updated. After a relatively few number of steps (as compared to the iterative method), the element control weight vector will be at its optimum value.

An iterative system may be characterized by a typical adaptive, tapped delay line equalizer used in data transmission. An example of a type of iterative algorithm which is used in a tapped delay line equalizer is the zero forcing type. This iteratively adjusts the equalizer tap gains to get a minimum voltage at the sampling instants. When this is achieved then one is assured of minimum intersymbol interference (ISI) and hence indirectly the best BER. An example of a recursive system is the Kalman filter. This algorithmic process is characterized by recursively calculating at each sampling instant an optimum weight control word.

The iterative algorithm asymptotically approaches an optimum solution after a series of iterative steps based on an indirect measurement. Whereas the recursive algorithm calculates the optimum solution at each step in the

process. The major implementation differences between the two families of algorithms are in the areas of speed of convergence, complexity, and overall performance. A summary of this is shown below in Table 1, for an N x N planar array.

Algorithm	Calculations Per Sample	Samples to Converge	Features
Iterative	Small (relative to N)	Large	Slow convergence simple implementation
Recursive	Large (relative to N^2)	Small	Fast convergence complex implementation

Table 1. Comparison of Features Between Iterative and Recursive Algorithms.

The iterative algorithm generally converges to the optimum solution relatively slowly. However it is quite simple compared to the recursive algorithm. The result is that iterative algorithms require very few computations between samples, however they require a relatively large number of samples (and iterations) before approaching an optimum condition. The recursive algorithm converges relatively quickly but is computation intensive compared to the iterative. The implication here is that while very few samples are required to converge, the time to converge may be long depending on the processor implementation, and processing speed. Therefore the implementation complexity of the recursive is greater than the iterative.

As the HF channel is time varying, showing both short term and long term effects, then one does not have the freedom to take an infinite number of samples (in the iterative case) or an infinite amount of processing time between samples (in the recursive case). Thus one of the main problems in selecting an algorithm is to ensure that neither the number of samples are prohibitive and would prevent convergence due to short term changes during sampling. Nor the processing time is prohibitive and prevent convergence due to short term changes during processing. The result is that due to the time varying nature of the HF channel and also the operating environment, one does not have any compelling reasons to select nor reject one family of algorithms over the other, and both were considered in the initial investigation.

2.2.4 ITERATIVE ALGORITHMS

Iterative algorithms refer to the family of algorithms which use an iterative estimation process to produce an optimum solution. This type of algorithm "improves" the control element weight with each iteration and eventually approaches optimality (asymptotically optimal). These algorithms include the gradient-based and random search algorithms.

2.2.4.1 Gradient-Based Algorithms

The use of gradient-based algorithms to control adaptive arrays is popular because of their simplicity and applicability to a wide class of problems. Using an iterative process, these algorithms adjust the array weights to optimize some performance criterion. When this criterion is a quadratic function of the weight settings, the performance measure can be visualized as a bowl-shaped surface. Gradient-based algorithms estimate the gradient at a point on this surface and update the weights in order to reach the "bottom of the bowl." Several gradient-based algorithms have been investigated. A representative gradient-based algorithm, the Widrow-Hoff LMS algorithm, is shown in the following section.

2.2.4.2 Widrow-Hoff LMS Algorithm

One of the most widely used gradient based algorithms is the Least Mean Squares type (LMS). It has been used in many adaptive applications, such as channel equalization, noise cancelling, and antenna arrays. It is popular because it uses a modest number of computations, it is easy to implement in hardware, and its operation is well understood. The LMS algorithm attempts to minimize the mean squared error (MSE) between a desired response (reference) and the actual array output. To illustrate an LMS algorithm, the following definitions and procedures are given:

Input Signal Vector

$$\underline{X}_k = \begin{bmatrix} x_1 \\ x_2 k \\ x_{3k} \\ : \\ x_{Nk} \end{bmatrix}$$

Element Weight Vector

$$\underline{W} = \begin{bmatrix} w_1 \\ w_2 \\ w_3 \\ : \\ w_N \end{bmatrix}$$

(Note: k represents a discrete sampling instant. N is an antenna element. In general, the signal vector and element control vector are complex signals.)

The output of the array is y_k where

$$y_k = \underline{X}_k^T \underline{W}_k = \underline{W}_k^T \underline{X}_k$$

The error, ε_k, is defined as the difference between a reference response (d_k) and the array output (y_k).

$$\varepsilon_k = d_k - y_k$$

The mean-squared value of this quantity is to be minimized.

A general expression for the mean square error (MSE) as a function of the weights can be found, assuming the input signal is a stationary random process.

$$\varepsilon_k^2 = d_k^2 - 2d_k \underline{X}_k^T \underline{W} + \underline{W}^T \underline{X}_k \underline{X}_k^T \underline{W}$$

$$E[\varepsilon_k^2] = E[d_k^2] - 2E[d_k \underline{X}_k^T] \underline{W} + \underline{W}^T E[\underline{X}_k \underline{X}_k^T] \underline{W}$$

Let the vector P be the cross correlation between the scalar d and the signal vector X, and let R be the input correlation matrix.

$$P = E \begin{bmatrix} d_k\, x_{1k} \\ d_k\, x_{2k} \\ d_k\, x_{3k} \\ \vdots \\ d_k\, x_{Nk} \end{bmatrix} \qquad R = E \begin{bmatrix} x_{1k}\, x_{1k} & x_{1k}\, x_{2k} & x_{1k}\, x_{3k} & \cdots & \cdots \\ x_{2k}\, x_{1k} & x_{2k}\, x_{2k} & x_{2k}\, x_{3k} & \cdots & \cdots \\ x_{3k}\, x_{1}k & x_{3k}\, x_{2k} & x_{3k}\, x_{3k} & \cdots & \cdots \\ \vdots & \vdots & \vdots & & \\ \cdots & \cdots & \cdots & x_{Nk}\, x_{Nk} & \end{bmatrix}$$

The mean-square error can then be expressed as:

$$E[\varepsilon_k^2] = E[d_k^2] - 2\underline{P}^T\,\underline{W} + \underline{W}^T\underline{RW}$$

The error is a quadratic function of the element weights. It can be thought of as a parabaloid-type performance surface. Adjusting the weights to minimize the error consists of descending along the surface in order to reach the "bottom of the bowl." Gradient methods are often used.

The gradient of the error function is defined as:

$$\nabla_k = \begin{matrix} \dfrac{\partial E(\varepsilon_k^2)}{\partial w_1} \\ \dfrac{\partial E(\varepsilon_k^2)}{\partial w_2} \\ \vdots \\ \dfrac{\partial E(\varepsilon_k^2)}{\partial w_N} \end{matrix} = -2P + 2RW$$

The optimal weight vector (W_{OPT}) is found by setting the gradient (∇) of the MSE function to zero. The result is then:

$$\underline{W}_{OPT} = \underline{R}^{-1}\underline{P}$$

This is the well known Weiner-Hopf equation. In general, the exact solution requires the explicit measurement of correlation functions, and matrix manipulation. However, the LMS algorithm allows one to asymptotically approach an optimal solution without having to perform these tasks. This is a great advantage as these tasks are computation intensive.

The LMS algorithm uses the method of steepest descent. Using this method, the updated weight vector is equal to the past weight vector plus a change proportional to the negative gradient.

$$\underline{W}_{k+1} = \underline{W}_k - \mu\underline{\nabla}_k$$

where μ is a constant that controls stability and convergence.

The LMS algorithm estimates the instantaneous gradient by assuming that ε_k^2, the square of a single error sample, is an estimate of the MSE. This quantity is then differentiated with respect to W.

$$\hat{\nabla}_k = \begin{bmatrix} \frac{\partial \varepsilon_k^2}{\partial w_1} \\ \frac{\partial \varepsilon_k^2}{\partial w_2} \\ \vdots \\ \frac{\partial \varepsilon_k^2}{\partial w_N} \end{bmatrix} = 2\varepsilon_k \begin{bmatrix} \frac{\partial \varepsilon_k}{\partial w_1} \\ \frac{\partial \varepsilon_k}{\partial w_2} \\ \vdots \\ \frac{\partial \varepsilon_k}{\partial w_N} \end{bmatrix} = -2\varepsilon_k \underline{X}_k^*$$

The update equation for the LMS algorithm can then be written as

$$\underline{W}_{k+1} = \underline{W}_k + 2\mu\varepsilon_k\underline{X}_k^*$$

A different gradient estimation algorithm is shown in the next section.

2.2.4.3 Differential Steepest Descent (DSD) Algorithm

The DSD algorithm, like the LMS algorithm, attempts to find an estimate of the mean squared error gradient in order to adjust the element weight settings. The two algorithms arrive at this estimate in different ways. The DSD algorithm directly measures the performance curve and approximates its derivative.

For a parabolic performance surface, the derivative can be estimated by slightly perturbing the weight vector from W(k) to W(k) + δ to W(k) - δ and then using symmetric differences to approximate the derivative given by

$$\frac{\overline{d(\varepsilon^2_k)}}{d\underline{w}} = \frac{\overline{\varepsilon^2}(W_k + \delta) - \overline{\varepsilon^2}(W_k - \delta)}{2\delta}$$

The weight update equation can then be expressed as

$$\underline{W}_{k+1} = \underline{W}_k - K_s \frac{\overline{d(\varepsilon^2_k)}}{d\underline{w}}$$

2.2.4.4 Howells-Applebaum Algorithm

The Howells-Applebaum algorithm was among the first to be developed and is widely used. Where the LMS and DSD algorithms require a reference signal, this algorithm does not. Instead, a beam steering vector, $\underline{b}$, is used. The purpose of the beam steering vector is to steer the recieve beam in a desired direction. It is assumed that the direction of arrival of the information bearing signal is known. The algorithm is characterized by two modes of operation: an adaptive mode when the signal is present and a quiescent mode when the signal is absent. When the signal is absent, the adaptive weights settle to values needed to point the beam in the desired direction. This is controlled by the steering vector. Many algorithms of this type have been shown to perform better when the signal is absent. For this reason, they are used almost exclusively in radar systems. The resulting performance of the Howells-Applebaum and LMS is similar.

The optimum weight vector for the Howells-Applebaum algorithm is given by:

$$\underline{W}_{opt} = \text{constant} \times \underline{R}^{-1} \times \underline{b}^*$$

Notice that the LMS and Howells-Applebaum algorithm yield the same weight vector solution provided that $\underline{b}^*$ is equal to $\underline{P}$. (See page 2-10, middle)

2.2.4.5 Random Search Algorithms

Using the LMS algorithm, the MSE criterion results in a quadratic performance measure. Since the performance measure is unimodal (no local minima), then gradient techniques can be used to locate the "bottom of the bowl." If the performance measure is unknown, or it contains local minima, gradient techniques must be used with extreme caution. In this case random search

algorithms may be advantageous. Random search algorithms possess global search capabilities and are not affected by local minima. They also require few computations and are easy to implement. Three random search algorithms are outlined below.

2.2.4.6 Linear Random Search (LRS) Algorithm

This algorithm begins by tentatively adding a random change to the element weight vector at the beginning of each iteration. The performance measure is then observed and a permanent adjustment is made that is proportional to the product of the tentative weight change and the performance measure change. This relation can be expressed as

$$\underline{W}_{k+1} = \underline{W}_k + K_s[J(\underline{W}_k + \underline{u}_k)]\underline{u}_K$$

where

$J(\underline{w}_k)$ is the estimated performance measure

K_S is a step size constant that controls stability

$\underline{u}_k$ is a random vector that affects stability and convergence

2.2.4.7 Accelerated Random Search (ARS) Algorithm

The accelerated random search algorithm changes the complex element weights within some range of values as shown below.

$$\underline{W}_{k=1} = \underline{W}_k + K_S(k)\ \Delta\underline{W}_k$$

where:

$K_S(k)$ is the step size

$\Delta\underline{W}_k$ is a random vector

The random vector is defined as:

$$\Delta W_i(k) = \cos\Theta_i + j\sin\Theta_i \quad \text{for } i = 1, 2, 3, \ldots, N$$

so that Θ_i is a uniformly distributed random angle between 0 and 2π. $\underline{\Delta W}_K$ controls the direction of the weight vector change and K_S controls the magnitude.

The weight vector is changed and a new performance measure is evaluated and compared to the previous estimate. If the new estimate shows improvement, the direction, ΔW_k , is retained and the step size is doubled. Otherwise, the previous weight vector is retained and the step size returns to its initial value.

2.2.4.8 Guided Accelerated Random Search Algorithm

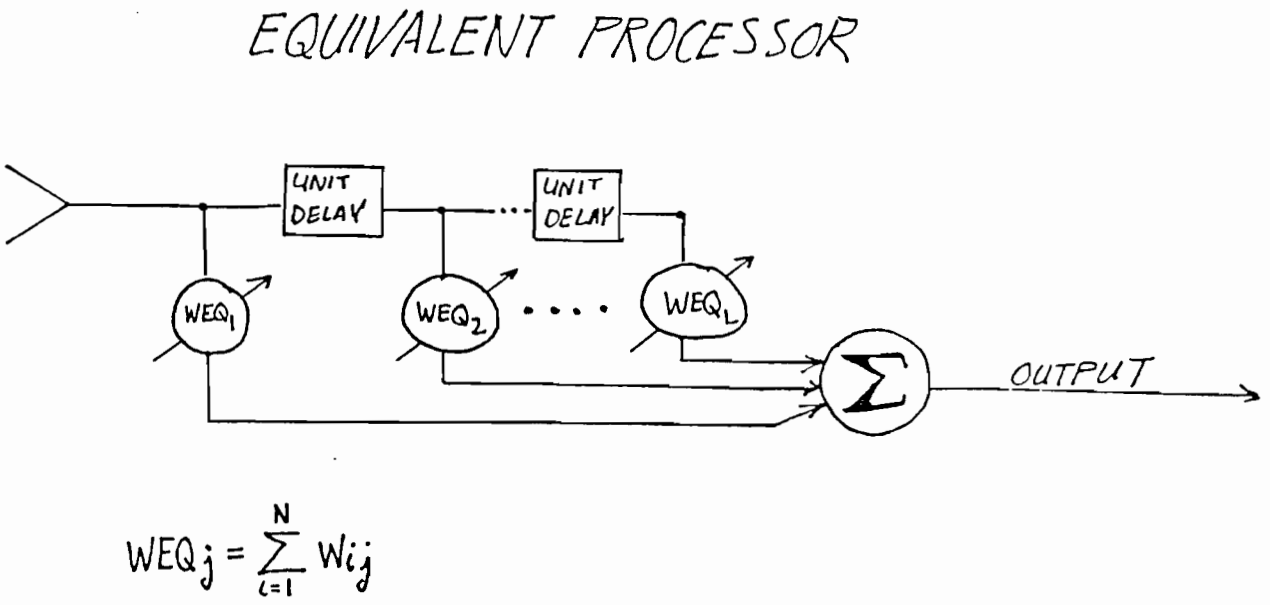
This algorithm has two stages, a random stage and a deterministic stage. When the weight vector is in a direction that produces improvement in the performance measure, control switches to the deterministic stage. Any time in the deterministic stage that an unsuccessful outcome results, control can be switched back to the random stage, depending on size of error.

2.2.4.9 Constrained LMS Algorithm

The last iterative algorithm investigated was the Constrained LMS algorithm. The name is somewhat misleading as the two algorithms (LMS and Constrained LMS) are quite the same. Some of their constrasting features are illustrated below.

Features	LMS	CONSTRAINED LMS
Reference Signal	Required	Not Required
Arrival Angle of Comm. Signal	Not Required	Required
Performance Criterion	M.S.E.	Maximum Likelihood
Optimization Technique	Minimize error	Minimize Output Power according to constraints

The array model that will be used to discuss this algorithm is shown in figure 2.2.4.9-1. Notice that ths model can accomodate broadband signals. When narrowband signals are used, a simplified model results, as will be discussed later. Also shown is an "equivalent processor" in figure 2.2.4.9-2. This model aids in the understanding of the Constrained LMS algorithm operation.



$$WEQ_j = \sum_{i=1}^{N} W_{ij}$$

Figure 2.2.4.9-1 Array Model

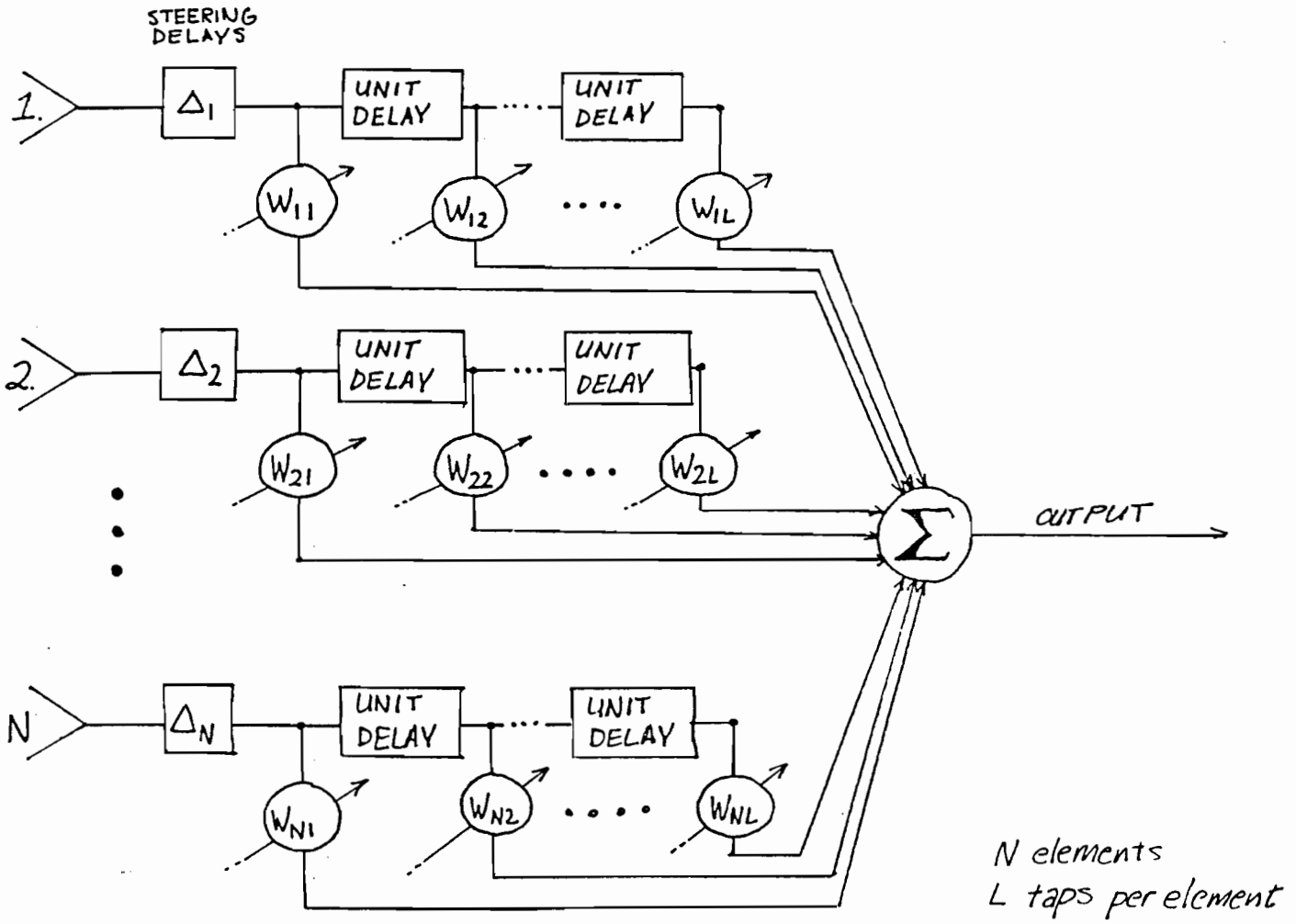


Figure 2.2.4.9-2 Equivalent Processor for Array

The function of the steering delays is to focus the array pattern in the direction of the desired signal. For a given arrival angle, these delays can be calculated from the array geometry. Once they are introduced, the signal arrives at the "same time" at each element as if the signal was broadside to a linear array. The desired signal component is identical at each element output.

At this point it is useful to examine the output of both the array model and the equivalent processor. The ouptut of the array can be expressed as

$$\begin{aligned} & X(k)\, W_{11} + X(k-1)\, W_{12} + \ldots X(k-L+1)\, W_{1L} \\ & + X(k)\, W_{21} + X(k-1)\, W_{22} + \ldots X(k-L+1)\, W_{2L} \\ & \quad \vdots \qquad\qquad \vdots \qquad\qquad\qquad \vdots \\ & + X(k)\, W_{N1} + X(k-1)\, W_{N2} + \ldots X(k-L+1)\, W_{NL} \\ & = X(k) \sum_{i=1}^{N} W_{i1} + X(k-1) \sum_{i=1}^{N} W_{i2} + \ldots X(k-L+1) \sum_{i=1}^{N} W_{iL} \end{aligned}$$

The output of the equivalent processor can be written as

$$X(k)\, WEQ_1 + X(k-1)\, WEQ_2 + \ldots X(k-L+1)\, WEQ_L$$

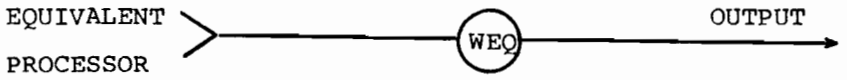
Therefore, if WEQ_j is chosen to equal $\sum_{i=1}^{N} Wij$, the outputs of the models are equivalent. This is an extremely important point.

By fixing the equivalent processor weights, it is possible to select a specific response in the look direction. The individual array weights can take on any value provided that the "sum on columns" constraint is satisfied. In other words, regardless of the values of the individual weights, if:

$$WEQ_j = \sum_{i=1}^{N} Wij$$

then the selected response in the desired direction is maintained. Therefore, it is possible to adjust the weights to null out interferers without degrading the response in the direction of the desired signal. It should also be noted that if the response selected is unity gain with zero phase delay in the desired direction, the output contains an undistorted version of the desired signal plus noise. The output is therefore the maximum-likelihood estimate of the signal.

The tapped delay line in the equivalent processor allows a user to select a specific frequency response. If no special filter is needed (as in the narrowband case), the model is greatly simplified. The tapped delay line can be replaced with a zero-phase, unity gain weight value.



WEQ = (1,0)

The constraint can then be stated as follows:

> The complex element weights can take on any values provided that their sum is equal to (1,0).

The algorithm itself will now be discussed.

The Constrained LMS algorithm consists of two "half-steps." The first half-step is expressed as

$$\bar{W}_{k+1/2} = \bar{W}_k + 2\, k_s\, \Delta t\, y_k\, \bar{X}_k^*$$

Notice that this equation is virtually identical to the LMS weight update equation except that the error signal is replaced with the array output, y_k. This is due to the fact that the constrained algorithm does not attempt to minimize error, but rather it minimizes the output power. Since the array pattern is constrained in the desired direction, there is no danger of minimizing the power from the desired signal.

The second half step consists of re-establishing the constraint. A constraint error is defined as

$$e_{k+1/2} = \left[(1,0) - \sum_{i=1}^{N} W_{i_{k+1/2}}\right]/N \quad ; \quad \bar{E}_{k+1/2} = \begin{bmatrix} e_{k+1/2} \\ \vdots \\ e_{k+1/2} \end{bmatrix}$$

The updated weights are then generated as

$$\bar{W}_{k=1} = \bar{W}_{k+1/2} + \bar{E}_{K+1/2}$$

2.2.5 Recursive Algorithms

Recall that the LMS algorithm uses gradient estimates in order to asymptotically arrive at an optimal weight solution, as shown in Weiner-Hopf relation:

$W = R^{-1} P$

W = Optimum weight solution

R^{-1} = Inverse sample covariance matrix

P = Cross correlation of desired response and signal vector

(This is the actual solution for algorithms that minimize MSE. LMS approximates this only.)

An alternative to both the iterative and recursive approaches could be to explicitly measure the correlation functions and perform a direct matrix inversion. This should result in better performance, since no crude estimates are involved in the calculation of the optimal element weights. However, in the past, this method was not feasible due to the number of computations required and processing speed available at that time. Advances in computer and device technology have now made this method faster. However for some applications, it is still not feasible due to processing time.

Recursive algorithms offer an alternative to the brute force methods of direct matrix inversion. These algorithms perform the inversion in a recursive manner. They are closely related to Kalman filtering techniques. Although these algorithms are still computation intensive, they offer marked improvement in convergence rates, given some fixed processing capability.

2.2.5.1 Weighted Least Mean Square Algorithm (WLMS)

The output of an N element array can be written as

$$y_{(t)} = \underline{w}(t)^T \underline{x}(t)$$

where $\underline{x}(t)$ is a vector of received signals and $\underline{w}(t)$ is a vector of multiplicative weights. The performance measure for the weighted least mean square (WLMS) can be expressed as

$$J(\underline{w}) = \tfrac{1}{2} \sum_{i=1}^{k} \alpha_i [\underline{w}^T \underline{x}(i) - d(i)]^2$$

where the α's are weights used to de-emphasize old data points. In matrix notation, this becomes

$$J(\underline{w}) = \tfrac{1}{2} [\underline{X}(k)\underline{w} - \underline{d}(k)]^T \underline{A}_k^{-1} [\underline{X}(k)\underline{w} - \underline{d}(k)]$$

where

$$\underline{X}(k) = \begin{bmatrix} x(1) \\ x(2) \\ : \\ : \\ x(k) \end{bmatrix} \qquad \underline{d}(k) = \begin{bmatrix} d(1) \\ d(2) \\ : \\ : \\ d(k) \end{bmatrix} \qquad \underline{A}_k = : \begin{bmatrix} \alpha^{k-1} & 0 & \cdots\cdots\cdots 0 \\ 0 & \alpha^{k-2} & \cdots\cdots\cdots 0 \\ : & & : \\ : & : & \alpha\cdots\cdots\cdots 0 \\ 0 & 0 & 0\cdots\cdots\cdots 1 \end{bmatrix}$$

The optimum weight selection is found by taking the derivative of J(w) with respect to w and setting the result to zero.

$$w_{\ell S}(k) = [\underline{X}^T(k)\underline{A}_K^{-1}\underline{X}(k)]^{-1} \underline{X}^T(k) \underline{A}_K^{-1} \underline{d}(k) \quad \text{and}$$

$$w_{\ell S}(k+1) = [\underline{X}^T(k+1)\underline{A}_{k+1}^{-1} \underline{X}(k+1)]^{-1} [\alpha \underline{X}^T(k)\underline{A}_k^{-1} d(k) + \underline{x}(k+1)\, d(k+1)]$$

$$\underline{P}^{-1}(k) = \underline{X}^T(k)\underline{A}_K^{-1}\underline{X}(k)$$

By defining $\underline{P}^{-1}(k)$ as above, a recursive relation is found for the weight update solution that does not require direct inversion. This is shown below.

$$\underline{w}(k+1) = \underline{w}(k) + \frac{\underline{P}(k)\, [\underline{x}^*(k+1)d(k+1) - \underline{x}(k+1)y(k+1)]}{\alpha + x^T(k+1)\underline{P}(k)x^*(k+1)}$$

where

$$\underline{P}(k+1) = \frac{1}{\alpha} P(k) - \frac{\underline{P}(k)\, x^*(k+1)\, \underline{x}^T(k+1)\, \underline{P}(k)}{\alpha + \underline{x}^T(k+1)\, \underline{P}(k)\, \underline{x}^*(k+1)}$$

2.2.5.2 Update Covariance Algorithm

The update covariance algorithm is very similar to the WLMS algorithm. In this algorithm, the sample covariance, R , is used to establish the effect of old data. The update equations of interest can be summarized as follows:

$$\hat{\underline{w}}(k+1) = \frac{\alpha k + 1}{\alpha k} \frac{\hat{\underline{w}}(k) - \hat{\underline{R}}_{xx}^{-1}(k)\, \underline{x}^*(k+1)\, \underline{x}^T(k+1)\, \hat{\underline{w}}(k)}{\alpha k - \underline{w}^T(k+1)\, \hat{\underline{R}}_{xx}^{-1}(k)\, \underline{x}^*(k+1)}$$

where

$$\hat{\underline{R}}_{xx}^{-1}(k+1) = \frac{\alpha k + 1}{\alpha k} \frac{\hat{\underline{R}}_{xx}^{-1}(k)\, \underline{x}^*(k+1)\, \underline{x}^T(k+1)\, \hat{\underline{R}}_{xx}^{-1}(k)}{\alpha k + \underline{x}^T(k+1)\, \hat{\underline{R}}_{xx}^{-1}(k)\, \underline{x}^*(k+1)}$$

2.2.5.3 Minimum Variance Algorithm

As the name indicates, the goal of this algorithm is to minimize the output noise variance. The performance measure can be written as

$$J = \underline{w}^T \underline{R}_{nn} \underline{w} \qquad \text{subject to the constraint}$$

$$\sum_{i=1}^{N} w_i = 1$$

The update equations of interest are given below.

$$\underline{w}_{mv}(k+1) = \frac{\hat{\underline{R}}_{xx}^{-1}(k+1)\, \underline{1}}{\underline{1}^T\, \hat{\underline{R}}_{xx}^{-1}(k+1)\, \underline{1}}$$

where

$$R(k+1) = \frac{\alpha k + 1}{\alpha k} \hat{\underline{R}}_{xx}^{-1}(k) - \frac{\hat{\underline{R}}_{xx}^{-1}(k+1)\, \underline{x}^*(k+1)\, \underline{x}^T(k+1)\, \hat{\underline{R}}_{xx}^{-1}(k)}{\alpha k + \underline{x}^T(k+1)\, \hat{\underline{R}}_{xx}^{-1}(k)\, \underline{x}^*(k+1)}$$

2.2.6 Summary of Adaptive Algorithms

This report introduces several types of adaptive algorithms. (It is certainly not all inclusive and we recoginize that many variations of these algorithms exist.) The following is a summary of the application and advantages of these algorithms.

- - The Widrow-Hoff LMS algorithm has become somewhat of a standard for adaptive algorithms. It is easy to implement and has a sufficient convergence rate for many applications.

- - The Differential Steepest Descent algorithm is similar to the LMS. However, it is less efficient and therefore rarely used.

- - The Howells-Applebaum algorithm is also a commonly used algorithm and was among the first to be developed. Its performance is very similar to that of the LMS algorithm. Due to the nature of the algorithm, it is particularly well suited to radar rather than communication applications. Also, it is very sensitive to steering errors.

- - The random search algorithms have the distinction of being the slowest algorithms discussed. This certainly detracts from their usefulness. Unless the performance measure is unknown, random search algorithms are not generally used.

- - The recursive algorithms offer the best performance in terms of convergence rates. They may offer large improvements over the iterative algorithms. Unfortunately, a heavy price is paid in terms of complexity. They also require a large number of computations at each sampling instant. For comparison, consider an N element array. A Kalman type algorithm requires about $5N^2$ computations/sample while the LMS requires about 2N computations/sample. As the number of antenna elements increase, the feasibility of the recursive algorithms decreases.

An adaptive array for HF systems, such as described in the RADC Statement of Work, Receiving System. . ., 8 January 85, presents some interesting problems from the algorithm standpoint. First, an array of 64 elements will

require a large amount of processing for each time sample. The algorithm must be capable of performing all necessary computations for each sample before the next sampling instant (for real-time processing). Secondly, the actual HF channel can have fast fading and changes in signal angles can occur relatively rapidly. The algorithm must converge fairly rapidly to be able to track signals in the channel. In other words, the algorithm must be both fast and simple. This is a difficult task.

At this time, the LMS-type algorithms appear to be the most promising. There has been some concern as to whether this type of algorithm is applicable for HF channels, but it is known that some have been proposed and successfully simulated (39, 40). The utilization of training sequences in a preamble can help to alleviate the slow convergence problem. The major advantage of this type of algorithm is the small computational load. Although the rapid convergence rates of recursive algorithms make them very attractive, the size of the proposed RADC array (or any practical array size) all but drops them from consideration. (A recursive algorithm operating on a 64 element array would require on the order of 20,000 complex computations for each sample.)

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