THE UNIVERSITY OF HULL

ENHANCED FREQUENCY MANAGEMENT FOR AUTOMATIC HF RADIO COMMUNICATION SYSTEMS

being a Thesis submitted for the Degree of

Doctor of Philosophy

in the University of Hull

by

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July 1989.

SUMMARY

The work described in this thesis aims to enhance the frequency management of automatic high frequency (HF) radio communication systems. During the research programme two new frequency management tools were developed; a chirpsounder monitoring tool to provide accuracy enhancement information for propagation prediction programs and an algorithm designed to allow optimisation of signal formats, so that in-band interference is avoided and the overall system throughput rate is increased. Two new HF communication system architectures are presented, which use system design and programming methodologies derived from the fields of artificial intelligence and computer networks.

The characteristics of the HF band are presented from a communicator's viewpoint, rather than the generalised, technical approach normally associated with such reviews. The methods employed by current HF communication systems to overcome the inherent time and frequency variability of HF channels are presented in the form of reviews of propagation, natural noise and co-channel interference prediction methods, embedded real-time channel evaluation algorithms and HF communications system architectures. The inadequacies of these current techniques are analysed. The eradication of their shortcomings is the main objective of the work described in the thesis.

The short-term inaccuracies associated with current propagation analysis procedures can limit the performance of automatic HF communication systems. An accuracy enhancement methodology is proposed which makes use of measurements made on oblique chirpsounder transmitters. In order to provide accuracy enhancement data, a chirpsounder-based, propagation monitor was constructed. Its implementation and trials are described and methods of using its output to enhance prediction model accuracy are discussed. Ways in which its performance may be improved are detailed.

The theory of a technique, termed "template correlation", which provides automatic HF communication systems with signal format adaptation data in order to enable them to avoid in-band interference, is presented. The objective of this work is to enhance the error-free capacity of a channel via adaptation of the signal. The results of computer simulations and laboratory bench trials of template correlation are presented. Enhancements of the technique in the light of the trials results are included.

Two proposed design methodologies for automatic HF communication systems are described. The first uses many of the frequency management tools associated with current automatic systems and it combines the information from these using a blackboard-based expert system architecture. The second proposed design is more conceptual than the first. An inductive expert system is employed to produce rules describing the ways in which an automatic HF system should respond to certain path conditions. Examples of how such a system might function are given.

The single, most important factor which has enabled the techniques described in this thesis to be feasible is the availability of cheap but powerful microprocessors. Thus the overall philosophy of the work is to improve the performance of automatic HF communication systems via the incorporation of processing power and "intelligent software" into the communication system's terminals.

CONTENTS

ABBREVIATIONS USED

ACF	:	Autocorrelation Function
AGC	:	Automatic Gain Control
AI	:	Artificial Intelligence
ARQ	:	Automatic Repeat-Request
BER	:	Bit Error Rate
CCF	:	Crosscorrelation Function
CW	:	Continuous Wave
EDC	:	Error Detection and Correction
ERP	:	Effective Radiated Power
FEK	:	Frequency Exchange Keying
FFT	:	Fast Fourier Transform
FOT	:	Frequence Optimum de Travail
FMS	:	Frequency Management System
FSK	:	Frequency Shift Keying
HF	:	High Frequency
IFRB	:	International Frequency Registration Board
ISO	:	International Standards Organisation
ITU	:	International Telecommunications Union
LSB	:	Lower Sideband
LUF	:	Lowest Usable Frequency
MUF	:	Maximum Usable Frequency
OSI	:	Open Systems Interconnection
OWF	:	Optimum Working Frequency
PAP	:	Propagation Analysis Procedure
PCA	:	Polar Cap Absorption
PCE	:	Polar Cap Event
PDS	:	Power Density Spectrum
RTCE	:	Real-Time Channel Evaluation

SID	:	Sudden Ionospheric Disturbance
SNR	:	Signal-to-Noise Ratio
SSN	:	Sun-Spot Number
USB	:	Upper Sideband
UT	:	Universal Time

1 SCOPE OF THE THESIS

This thesis covers work performed by the author, as a member of the Hull-Warwick Communications Research Group (University of Hull), in conjunction with Plessey Research and Technology (Roke Manor) Limited between the dates of 1 October 1986 and 30 September 1989. The broad aim of the research was to enhance current frequency management procedures for automatic high frequency (HF) communication systems, in order to make the HF band more readily available to a greater number of users. This was to involve the development of new architectures and techniques for such systems.

Plessey had already developed an automatic HF system and this was to be used as a starting point for the work. A compatible terminal was to be set up at the University of Hull in order to allow trials to take place.

The structure of the thesis is as follows :

Chapter 2 - HF Radio Communications - A Users Perspective.
This covers the characteristics of the HF band from a user's point of view. The likely propagation, noise and interference conditions for a typical HF channel are presented, along with their effects on communications traffic.

Chapter 3 - Modelling HF Communications Path Characteristics.
Methods employed by current automatic HF communications systems to predict and model the behaviour of specific HF radio paths are presented. Sources of propagation, natural noise and interference data are discussed, and the inadequacies of the various prediction procedures are presented.

Chapter 4 - Embedded Real-Time Channel Evaluation (RTCE) Techniques.

A definition of RTCE is given and examples of embedded RTCE routines are presented, ie those routines which utilise the terminal equipment ordinarily used for communications purposes to derive path performance parameters. Passive and active RTCE algorithms are presented.

Chapter 5 - Overview of Current Automatic HF Communication Systems.

The rationale for automating the control of HF communication systems is presented. The software and hardware components of the Plessey automatic HF communications system are described and examples of other systems which have been implemented are given.

Chapter 6 - Aims of the Research Programme

The contents of Chapters 3, 4, and 5 are used to identify topics requiring further work. Enhancement of propagation analysis procedure output, extraction of HF channel interference information and the development of automatic HF communication system architectures and design methodologies were the specified aims of this research programme.

Chapter 7 - Propagation Models and Methods of Enhancing their Accuracy.

The inadequacies of propagation models are stated in Chapter 3. A chirpsounder-based propagation estimation system is described, the aim being to use the output of this device to provide propagation prediction model accuracy enhancement information. Improvements to the system are suggested and ways in which the data from it may be used to improve propagation model accuracy are presented.

Chapter 8 - Template Correlation.

Template correlation is a deterministic technique which enables signal formats to be optimised to suit the prevailing channel noise and interference characteristics. As such it provides frequency management routines, embedded within a communications terminal, with information regarding the current interference structure on a particular channel and it enables the signal format to be adjusted to minimise the bit error rate (BER) or alternatively to maximise the channel capacity. The theory of the technique is presented along with the results of some computer-based simulations. Details of bench-trials and possible enhancements of the techniques are included.

Chapter 9 - A New, Automatic HF Communication Systems Architecture.

A systems architecture is described which incorporates the frequency management tools presented in earlier chapters of the thesis. The overall operation of the terminal design is broken down into two sub-systems: the frequency management system and the data management system. The frequency management system employs artificial intelligence programming techniques, in the form of a blackboard-based architecture. The data management rountines are implemented according to the International Standards Organisation Open Systems Interconnection (ISO-OSI) 7-layer model.

Chapter 10 - An Alternative Automatic HF Communications System Architecture

An architecture is described which is radically different from that of current HF communications systems. It may not necessarily require the data provided by "traditional" frequency management tools for successful operation. The suggested design is based upon a learn-by-induction expert system. It would initially be fed example operational data from a current, successful HF communications system and it would use this data to induce rules concerning the behaviour of the overall system. These, along with other manually generated rules, would then be used to control the new system, and their data requirements would dictate the system's hardware and software configuration.

Chapter 11 - Future Work.

The aims of this research programme and the progress made on it are used to identify a programme of future work.

Chapter 12, References.

Chapter 13, Acknowledgements.

Chapter 14, Published Papers.

2 THE HF BAND - A USER'S PERSPECTIVE

2.1 Introduction

The HF band of the radio spectrum exhibits a unique combination of propagation, natural noise and interference conditions. Because of the inherent variability of the band, effective use of the medium for communication purposes presents an engineering challenge.

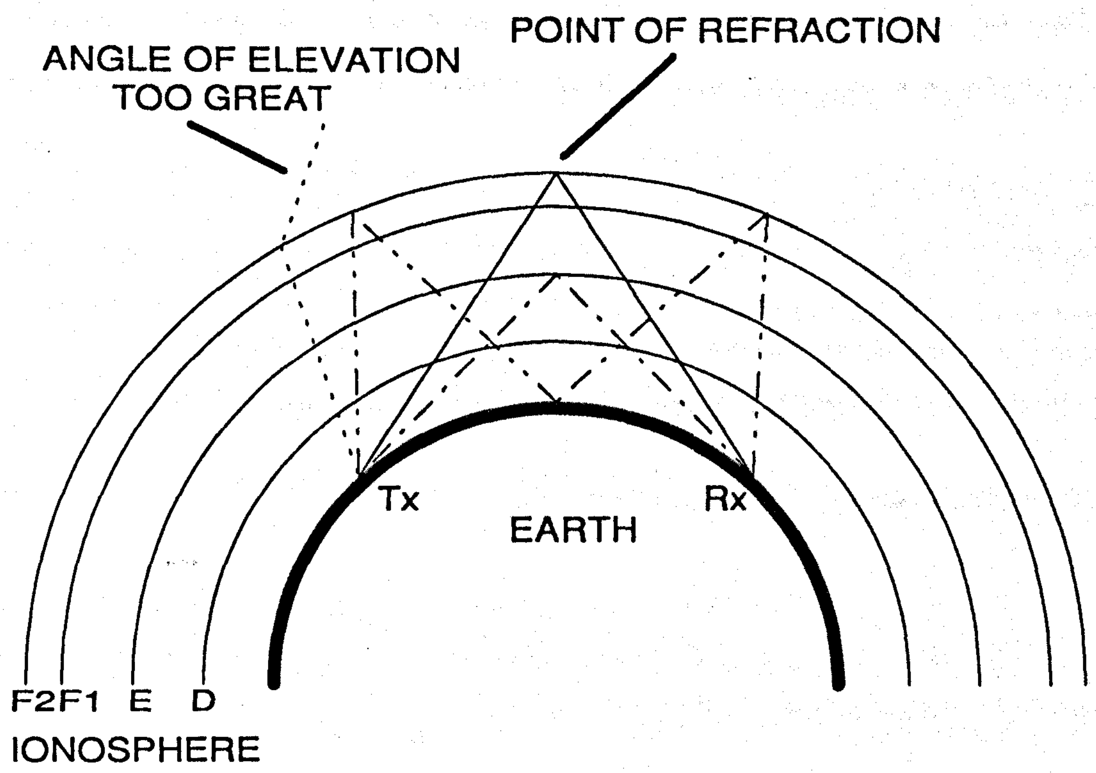
In this chapter the characteristics of the HF band are presented. The likely propagation, noise and interference conditions that an HF user may experience are discussed and their effects on communications traffic are presented. Finally the combined nature of these effects are examined in order to give an overall description of a typical HF channel from the user's point of view, as opposed to the more conventional, and purely technical, description.

2.2 Propagation Mechanisms

The HF, or Short-Wave, band of the radio spectrum occupies the region between 2 and 30 MHz; quarter wavelengths for the band are thus in the range 37.5m to 2.5m. These dimensions therefore represent the range of lengths of simple wire antennas that are resonant in the band. Hence, it is feasible for efficient HF antennas to be mounted on mobile platforms, such as ships, aircraft and man-packs, in addition to their being installed at fixed sites.

The two major propagation mechanisms that exist in the band are known as ground-wave and sky-wave. The engineering aspects of ground-wave propagation are in many respects the same as those for any other line-of-sight radio link; only

FIGURE 2.1. IONOSPHERIC PROPAGATION



sky-wave propagation is considered here since beyond line-of-sight communication is the primary concern of this research programme.

HF sky-wave propagation relies upon the refraction of radio wave energy by an ionised region above the earth's surface known as the ionosphere. Using this refraction mechanism over-the-horizon (OTH) propagation can be achieved, allowing trans-global communication to take place; this is illustrated by Figure 2.1.

The ionosphere consists of four main layers. The effects of each layer on incident HF radio energy and their approximate heights above the earth are shown below (Davies, 1966).

D-Layer	:	Absorber (80 km).
E-layer	:	Refractor and absorber (110 km).
F_1-layer	:	Refractor and absorber (190 km).
F_2-layer	:	Refractor (300km).

As can be seen from Figure 2.1, there can be more than one propagating mode between a transmitter and receiver. Multi-path conditions are thus common on HF links resulting in several versions of the transmitted signal appearing at the receiver with different time delays and signal strengths. Multi-path conditions limit the available channel capacity as the maximum data rate that can be reliably decoded at the receiver is given by the reciprocal of the total multi-path time delay (Davies, 1966). Generally, HF channels at the lower end of the band exhibit greater multi-path conditions, both in terms of the number of received signal components and the total multi-path time delay.

The component layers of the ionosphere vary according to time of day, season, position etc. During the day all four layers may be present. At night the F_2 and F_1 layers tend to merge and the D and E-layers often disappear; also, the remaining layers are much lower in altitude. Because of this change in constitution, a communicator using an HF channel over a 24-hour period will notice significant diurnal variations in received signal strengths and available channel capacity due to multi-path conditions.

During the day, the energy from the sun preserves the ionised state of the ionospheric layers since the amount of solar energy incident on the layer directly controls its ionisation density. At night, the ionosphere is preserved by the prescence of solar winds, ie particle radiation. The denser the concentration of ionisation in a layer, the greater is its refracting ability. Factors affecting the quantity of incident solar energy and particles are solar activity (usually specified by the sun-spot number (SSN)), the longitude and latitude of the refraction, the time of day and the time of year. It should be noted that the SSN exhibits a cyclic behaviour over a well-defined period of approximately 11 years.

Higher radio wave frequencies require a more highly ionised ionosphere in order for enough refraction to take place to return the wave to earth. Therefore for any sky-wave path there is a maximum usable frequency (MUF) which is directly governed by the ionisation density of the refracting layer. MUF is given by equation 2.1 below:

$$MUF = k.f_c.\sec\Phi_i \qquad (2.1)$$

where k is a factor to cater for the effects of the curvature of the earth, f_c is the vertical incidence critical frequency for the layer and Φ_i is the angle of incidence of the ray on the layer.

It is thus evident that the MUF for a particular path will vary with respect to time, in accordance with the factors governing ionisation density. As a result of this the MUF tends to be higher by day than by night (although under certain conditions, especially in summer, the day-time MUF can be lower than that experienced at night).

If the frequency of transmission f is held constant, and the angle of elevation of the launch of a ray of HF radio energy is increased, then a certain elevation angle is reached when the energy cannot be returned to earth by the refracting layer, if f is greater than the MUF for the layer. The ionisation concentration is then insufficient to provide the amount of refraction required, as illustrated by the dotted ray in Figure 2.1. There is thus an area of terrain around the transmitter which cannot be reached using sky-wave propagation at this frequency. This is known as the "skip-zone". Lowering the frequency of transmission can reduce the size of the skip-zone. Also, ground-wave propagation may be used to reduce the size of the skip zone.

The lowest usable frequency (LUF) for a path depends upon the levels of ionospheric absorption, the background noise level (see Section 2.3) and the communication system parameters. The D-layer is the principal absorber of HF energy and its absorption is greatest at the low frequency end of the band (Davies, 1966). Radio noise is also greatest at the lower end of the HF band. The highest level of D-layer absorption and radio noise that a particular HF communications system can successfully operate with will be governed by its transmitter power and antenna radiation patterns Hence the LUF for a particular path will vary between different HF communication systems.

Care is thus needed when choosing an operational frequency (and when selecting a complement of frequency allocations) in order to ensure that the temporal and geographical coverage achievable is adequate. Generally, in order for a user to be successful in achieving 24 hour per day coverage on a certain path, a frequency-agile communications system is required having frequency allocations spread over a wide range of the HF spectrum.

In addition to the ionisation density related effects mentioned above, there are a number of relatively unpredictable irregularities associated with an ionospheric radio link (Thrane, 1986). The following effects can occur in an undisturbed ionosphere, i.e. one where no solar events have influenced its composition.

i) **Fading.**

Rapid fades, with periods of a few seconds, can be caused by irregularities in the ionisation density of a layer (Davies, 1966). Such conditions cause the components of a signal to be refracted by differing amounts and there is thus potential for them to interfere with each other.

Fading can also be caused by the focussing and de-focussing of a beam of radio energy by an ionospheric layer (period = 15 - 30 minutes), the time variations in ionospheric absorption (period = 1 - 2 hours) and by the polarisation rotation of rays travelling through a layer (period 1 - 2 hours). HF paths also exhibit frequency selective fading, due to multi-path components of a signal interfering with each other.

ii) **Sporadic-E.**

This refers to the appearance of patches of ionisation at E-layer heights with an ionisation density greater than that of the normal E-layer. This can raise the MUF for a path and also it can mask the F_1 and F_2 layers, causing loss of communications to occur in the worst case.

iii) **Ionospheric Scatter.**

This gives rise to what is known as "spread-F" conditions. Irregularities in the ionisation density of a layer can cause the broadening of pulses of incident RF energy. This effect is normally defined in terms of its appearance on an ionogram and it can lead to difficulties in defining the

MUF for a path.

Other effects include ground-scatter, ionospheric tilts (causing off-great circle propagation and, in certain cases, an elevation of the MUF) and non-linear effects such as cross-modulation.

The ionosphere exhibits additional characteristics when so-called geophysical disturbances occur. These are outlined below.

i) **Sudden Ionospheric Disturbances.**

Solar flares result in a dramatic increase in the electron density of the lower ionospheric layers. This increase in concentration can cause a complete communications "black-out" due to the enhanced absorbing power of the layers; the resulting phenomenon is known as a sudden ionospheric disturbance (SID).

ii) **Magnetic Storms.**

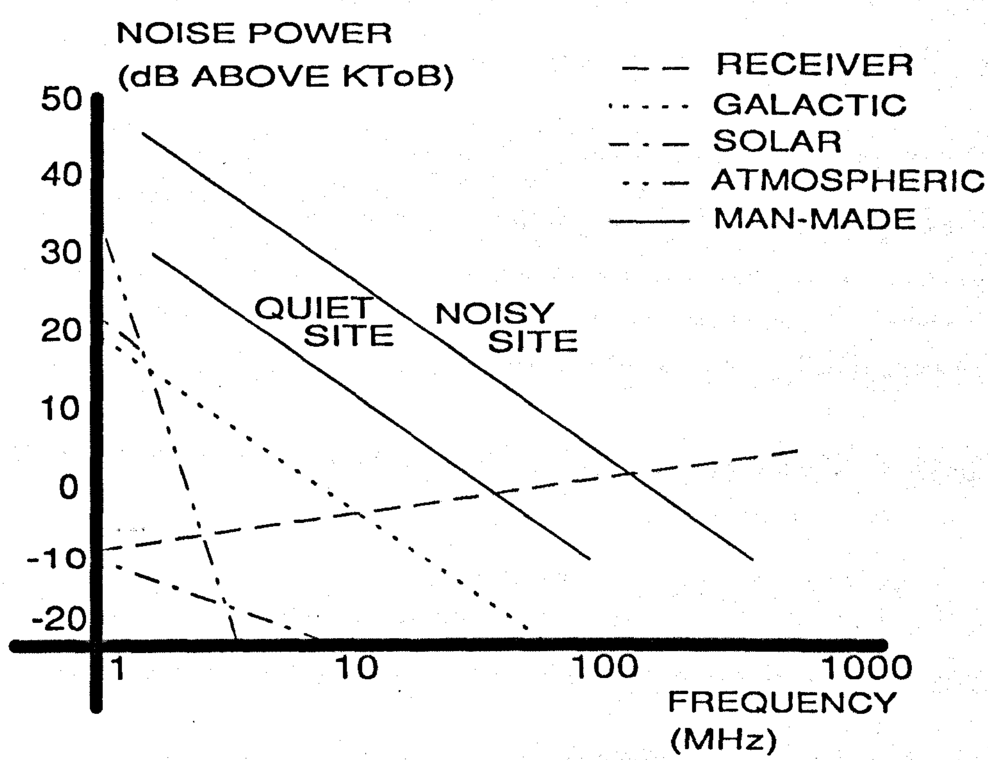
Magnetic storms around the earth can lead to strong and erratic time variations in the MUF. This effect is strongest and most likely in latitudes greater than 40° to 50°.

iii) **Absorption Anomaly.**

There is an effect known as the "winter anomaly" in ionospheric absorption. In mid-latitudes absorption may be increased by up to 60 dB over the normal value for an HF circuit, which can dramatically limit communications ability. This effect is thought to be due to particle precipitation from the sun and also to certain atmospheric events.

High latitude paths exhibit additional characteristics, such as polar cap absorption (Thrane, 1986).

FIGURE 2.2. AVERAGE NOISE POWERS FROM VARIOUS SOURCES



2.3 The Noise Environment In The HF Band

A plot of the powers of various radio noise sources as a function of frequency is shown in Figure 2.2. Receiver noise, i.e. that which is generated thermally by the electronic components in a radio receiver, is not significant in the HF band when compared with man-made (that from electrical machinery, ignition systems etc.) and atmospheric noise. Thus HF radio equipment is relatively easy to design from a noise figure point of view, in comparison with other higher frequency, long-range communications media, e.g satellites.

It is seen that, with the exception of thermal sources, noise power tends to decrease with increasing frequency. The least noisy channels tend to be at the higher HF frequencies (CCIR, 1963).

The co-channel noise experienced in the HF band is not usually Gaussian in nature (especially when interference is also present). It is important to note that since the performance ranking characteristics of most modulation schemes are evaluated under Gaussian noise conditions, it is possible that these may not be valid for a typical, non-Gaussian HF channel.

2.4 The Interference Environment In The HF Band

The term "interference" is used here to mean unwanted signal energy present in a user's allocated channels. The allocation of the radio spectrum is carried out on a world-wide basis by the International Telecommunications Union (ITU) in Geneva. The spectrum is wholly allocated in each of the different frequency assignment regions. Each user is normally allocated a selection of 3 kHz-wide channels, according to his communication requirements and the availability of the required portions of the spectrum. In Section 2.2 the mechanisms by which global propagation can be achieved were discussed. These same mechanisms also allow interference to occur on a global scale.

Non-linear devices, e.g. mixers, diodes, rusty metal structures, will also add to interference problems. Such devices create inter-modulation products (IP's) and harmonics which are radiated across the band, thus increasing the background noise level.

Work has been performed at the University of Manchester Institute of Science and Technology (UMIST) on characterising HF co-channel interference. This has shown that the probability of finding a 3 kHz-wide HF channel completely free of interference can be very low (Gott et al, 1982). The situation is made worse at night because of the disappearance of the HF absorbing layers in the ionosphere (Section 2.2). Interference can be the factor limiting HF communications system performance in spectrally-congested regions, such as Western Europe.

2.5 The Combined Effects Of HF Propagation, Noise And Interference Conditions

The effects of the condition described in Sections 2.2, 2.3 and 2.4 can be described in terms of an error-free channel capacity variation. Channel capacity is given by (Schwartz, 1981):

$$\text{Capacity} = B.\log_2(1 + SNR) \quad \text{bits/sec.} \qquad (2.2)$$

where B is the bandwidth of the channel and SNR is the signal-to-noise ratio within the channel bandwidth. B is usually limited by the multi-path conditions occurring on the channel. SNR is affected by the transmitter power, the system's antenna gains, the path length and the noise and interference environment within

FIGURE 2.3. VARIATION OF CHANNEL CAPACITY WITH TIME

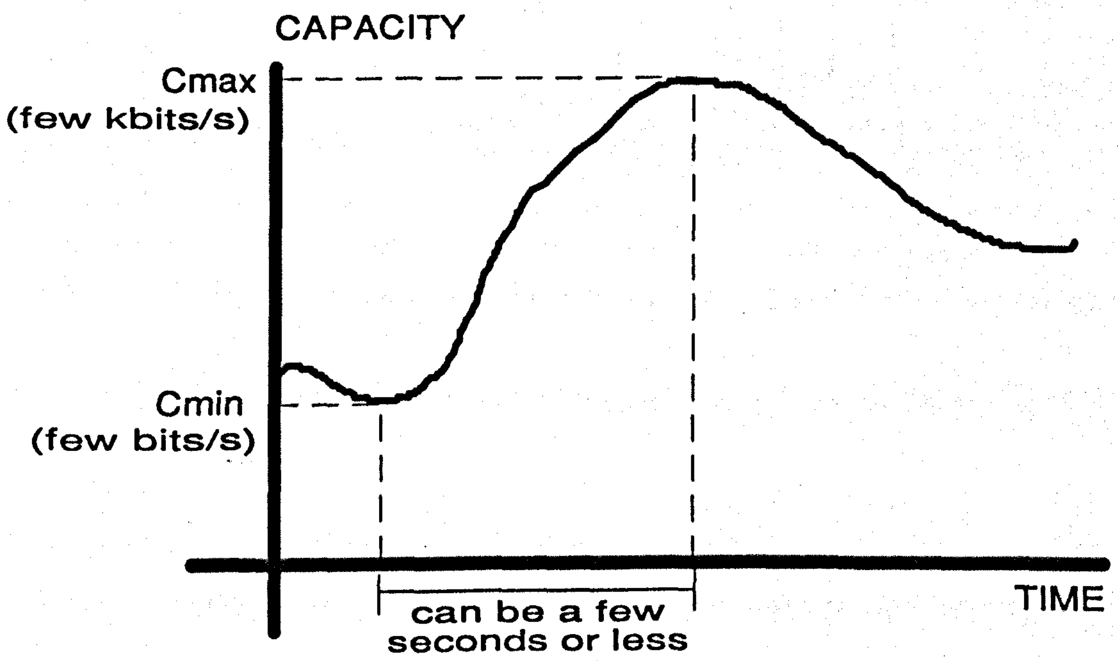


FIGURE 2.4 HF CHANNEL ERROR CHARACTERISTICS

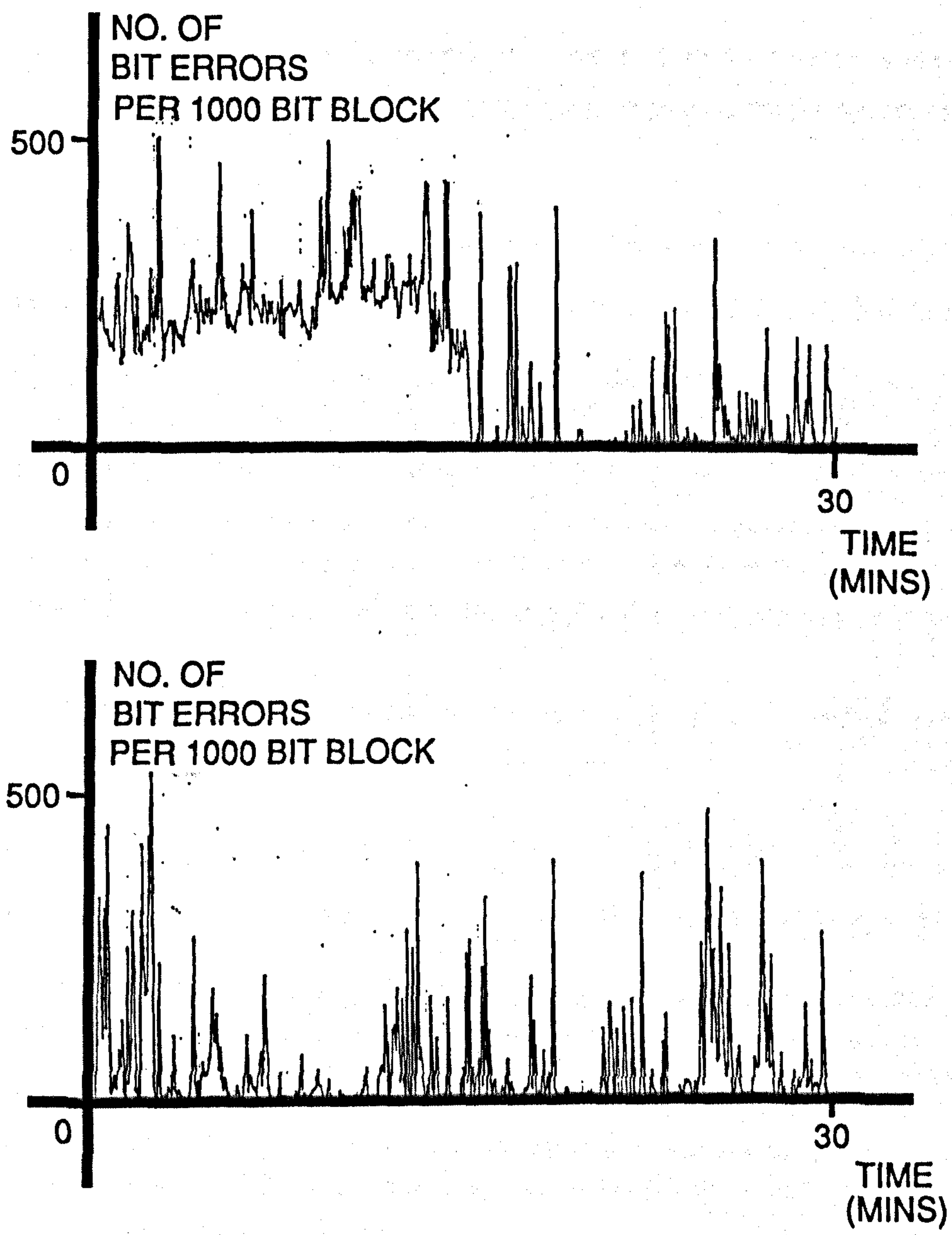
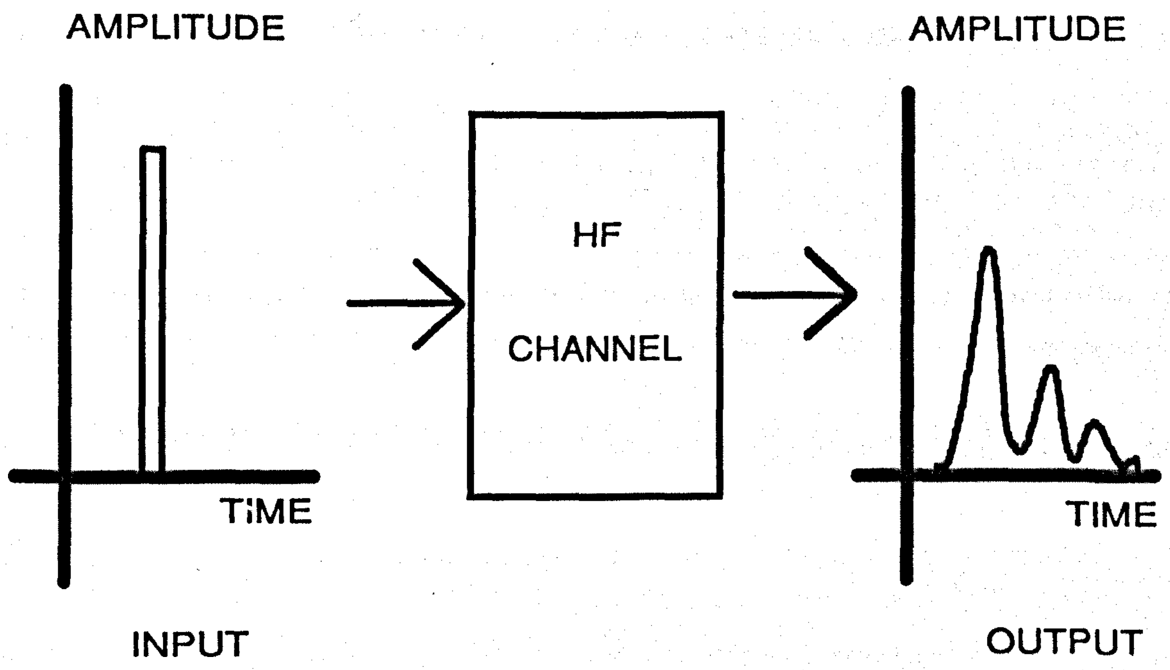


FIGURE 2.5. EFFECTS OF HF CHANNEL ON A SINGLE PULSE OF RF ENERGY



the channel bandwidth.

Figure 2.3 shows how this capacity can vary with respect to time (Darnell, 1986). The fluctuations in capacity can be very rapid with changes from a few kbits/sec to a few tens of bits/sec occurring in a few seconds or less.

The time variability of the quality of an HF channel is illustrated further by the error rate plots shown in Figure 2.4. Error rate and capacity are to some extent linked for a given transmission rate, as a channel with high capacity tends to have a low error rate (the converse also being true). These plots (supplied by Prof. M. Darnell, University of Hull) represent the number of errors occuring per 1000 bit block over a total time period of one hour. It can be seen that the channel changes rapidly from being totally corrupted to being error free, and vice versa. The intermittent nature of HF channel errors is well demonstrated by these results.

The possible effects on a single pulse of RF energy transmitted over an HF channel are shown by Figure 2.5. The return shows there to be multi-path components present, due to the existence of multiple propagation modes. The components are of differing strengths due to the unequal path attenuations of each of the modes. Also each mode has suffered dispersion. This occurs because different components of the signal are refracted by differing amounts, resulting from the "spreading" of the transmitted signal energy over a significant area of each ionospheric layer. This results in pulse broadening at the receiver.

The strengths and weaknessess of the HF band are described in (Darnell, 1986). It is a relatively cheap means of achieving long-distance communications. Simple equipment, low transmitter powers and simple antennas allow the medium to be accessed by mobile terminals. The long-term characteristics of the band are predictable with a fair degree of accuracy.

However an HF channel can be unpredictable and bursty in nature with the short-term behaviour differing greatly from the long-term. Levels of man-made interference can be high, especially at night. For digital encoding schemes, the medium can only support low average data rates (a few hundreds of bits/sec) and it is therefore only suitable for certain types of transmission, e.g. low-rate telegraphy, buffered high speed comunications. In order to use the medium effectively, an adaptive, frequency-agile communications system is required.

A communicator wishing to take advantage of the strengths of the medium must be aware of its pitfalls. He needs to be able to predict the conditions on a channel with a fair degree of accuracy. There are available computer-based prediction algorithms which produce path parameters for an HF radio link. These programs remove some of the uncertainty associated with the use of HF sky-wave communications and they are described in the next chapter.

Generally, a communicator wishing to make use of the HF band will have certain requirements for the transmission of the message. These may include the length of message, the destination, the latest tolerable arrival time and the security level of the transmission. These factors, coupled with the working parameters of the communications system itself, such as receiver and modem sensitivities, the transmitter power and the antenna gains, will decide which of the user's frequency allocations is optimum.

Given the requirements of the user, the operational parameters for the communications system need to be derived. In particular, accurate information is needed about the ranking of the frequency allocations according to their ability to pass the required traffic, the transmitter power necessary, the optimum transmission format (to give the required data rate and to avoid interference) and the expected time interval before any significant changes will occur on the link.

It is the task of the frequency management component of an HF system (whether automatic or otherwise) to provide such information in response to the requirements of the user. The more accurate the frequency management information the more successful will be the communications over the chosen path. The provision of effective and efficient HF frequency management procedures is thus the main aim of the work described in this thesis. Essentially, the major problem associated with efficient HF communication system control is a significant "mismatch" between on the one hand, the way in which user requirements are specified and the manner in which HF system parameters are monitored and adapted on the other. This research seeks to reduce that mismatch.

The following three chapters review the starting points for the work in terms of:

(i) the status of propagation analysis,

(ii) the range of appropriate RTCE techniques available, and

(iii) the current state-of-the art in automatic HF system design.

3 MODELLING HF COMMUNICATIONS PATH CHARACTERISTICS

3.1 Introduction

The previous chapter described the HF communications medium from a user's point of view. It was shown that the parameters of a typical HF path, such as available capacity, noise levels etc., can be extremely variable with respect to time and frequency of operation. In order to achieve successful communications efficient and effective HF system control algorithms are required with the ability to determine the optimum channel frequency for a particular path at any one time.

The methods used to derive channel parameters for HF communication systems fall into two main categories:

(i) models and prediction algorithms;

(ii) RTCE techniques.

HF channel models and prediction routines are used before a communications link is established, ie they assist the frequency management system (FMS) in determining the optimum channel for the path in question before any traffic is passed.

RTCE routines (covered in Chapter 4 of this thesis) are a means of path parameter characterisation via the use of probing signals and the actual communications traffic (Darnell, 1986). RTCE information can thus be extracted both before and during message transmission.

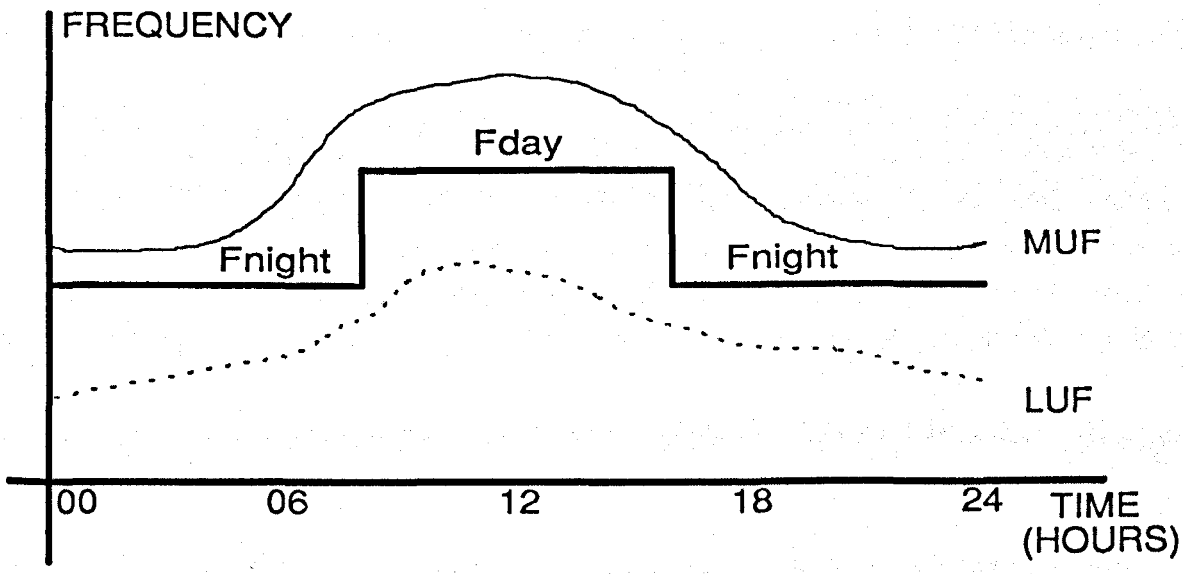
From a communications systems point of view, the models and prediction routines are required to provide propagation, natural noise and interference data, both during the design and operational stages of a system's life-cycle.

During the design of an HF system, it is advantageous to model the range of conditions that are likely to be encountered by all its terminals over the whole system life. This will give an indication as to which operational frequency assignments are the most desirable. If these assignments are then allocated to the system, the potential availability and reliability of communications links within the system's network of terminals should be maximised.

Using models and prediction routines, it is possible to assess the likely signal-to-noise ratios (SNR's) for given paths with different antennas and transmitters, thus enabling the optimum terminal equipment to be selected. It is cheaper and quicker (but less accurate) to run HF path analysis routines many times than to embark upon a measurement campaign in order to characterise the behaviour of the path. Hence, the suitability of different terminal equipment configurations for use on a particular path can be tested, whilst incurring a relatively low-cost.

When an HF system is operational, the models and prediction algorithms are used to try and decide which, out of all the user's frequency allocations, is likely to be optimum before communications traffic is passed. The aim is to try and produce a frequency operating schedule, such as the simple two-frequency schedule shown by Figure 3.1. A typical system will have more than two frequencies allocated to it and thus the frequency operating schedule may be considerably more complex than that shown by Figure 3.1.

FIGURE 3.1. EXAMPLE OF FREQUENCY PLANNING



3.2 Data Sources Used In The Production Of HF Propagation Analysis Procedures And Models

Propagation analysis procedures and models have evolved over many years by making use of a progressively more precise physical descriptions of propagation mechanisms and more comprehensive databases of propagation measurements; these continue to be augmented and refined.

The main international bodies responsible for the gathering of propagation data are (Thrane, 1986):

(i) International Scientific Radio Union (URSI).

(ii) International Radio Consultative Committee (CCIR).

(iii) Inter-Union Commission for Solar-Terrestrial Physics (IUCSTP).

In addition to the above-named organisations, many national agencies are involved in the collection of ionospheric propagation data.

The techniques used to gather the information for the database are:

(i) Vertical incidence sounding, eg using an ionosonde, which produces an ionogram as its output.

(ii) Oblique incidence sounding.

(iii) Backscatter sounding, which may be used to sound areas of the ionosphere which are difficult to reach using methods (i) or (ii) (eg oceanic or polar regions).

(iv) Satellite mounted "top-side" sounding.

(v) Measurement of the amplitude of VHF galactic radio noise at the earth's surface using a riometer. This allows significant changes in D-layer absorption to be detected.

(vi) Measurement of the variations in the earth's magnetic field using a magnetometer.

(vii) Rocket-mounted data collection via measurements made in transit through the various ionospheric layers.

(viii) Monitoring of identifiable transmissions, such as standard frequency and time signals, by calibrated signal strength receivers.

The basic information extracted from the above techniques is processed and then it is used to augment the database from which propagation analysis procedures are constructed.

3.3 Propagation Analysis Procedures

3.3.1 Introduction

Propagation analysis procedures (PAPs) are produced via two distinct methodologies:

(i) Use of the extensive database of propagation measurements, mentioned in Section 3.2 above, to derive empirical formulae to describe the likely behaviour of a path.

(ii) Use of the above database and an attempt to model the ionosphere more precisely in terms of deterministic expressions and parameters describing, in a more direct fashion, the physical processes occuring within the ionosphere, eg layer height, signal absorption levels, solar activity etc.

Method (i) works reasonably well if the paths of interest are well represented by the measurements in the database. However, the empirical formulae produced may not be directly related to the physical processes occurring within the ionosphere. Because of this, method (ii) is generally more comprehensive and it is the one which is the most widely used in HF PAPs. The methods are similar in that they try to predict future trends for a path from past characteristics.

In general, PAPs give as their output long-term characteristics for a particular path, such as monthly median figures for the MUF. For the purposes of an automatic HF FMS, embedded within the overall communications system architecture, PAP output is used in conjunction with RTCE routines, in order to gain both long and short-term information concerning the behaviour of a path - as discussed in Chapters 4 and 5.

Many PAPs are very large computer programs which can only be run using the hardware support offered by mainframe computers. From a communication systems viewpoint, smaller programs, which can be embedded within each terminal of the HF system, are of more use. Considerable research effort has thus been spent in producing compact algorithms. One such procedure, known as MICROP-3 is described in (Dick & Lewis, 1989).

Typical input information to a PAP is:

(i) Transmitter and receiver location.

(ii) Antenna type.

(iii) Required SNR at the receiver, and whether diversity processing is available.

(iv) The noise environment at the receiver, eg quiet rural, residential, business. Specific receiving site noise measurements can often be included into the program.

(v) The available frequency allocations.

(vi) Solar activity, in the form of sun–spot number or solar flux.

(vii) Time of day.

(viii) Month.

(ix) The ground constants for the transmitting and receiving sites.

(x) Available transmitter power.

In order to produce path performance parameters, PAPs normally perform the following general types of calculation: (Thrane, 1983):

(i) Calculation of MUF, including the effects of sporadic–E layer propagation.

(ii) Calculation of the transmission loss between the transmitter and receiver, for the given transmitter power and antenna types. This gives a figure for the field strength at the receiver.

(iii) Calculation of the SNR at the receiver by consideration of the natural and man-made noise sources and (ii) above.

(iv) Evaluation of the effects of the statistical variations in the ionospheric and noise parameters on the overall prediction results.

Details of the CCIR recommended methods for performing the above calculations can be found in (CCIR, 1970) and (CCIR, 1978).

3.3.2 Examples of PAP output

Figures 3.2 – 3.4 are example outputs from the HFMUFES4 PAP (Haydon et al, 1976). This program makes use of the analysis methods presented in CCIR report 252-2 (CCIR, 1970) and it is typical, in many respects, of a main-frame based PAP.

Figure 3.2 is a graphical output showing the variation of MUF, frequence optimum de travail (FOT) and lowest usable frequency (LUF) as a function of universal time (UT).

Figure 3.3 is an output format which gives the path reliability with respect to frequency and universal time.

Figure 3.4 shows part of a tabular output depicting detailed information about a specific path with respect to universal time and frequency.

The definitions of the symbols used in Figures 3.2 – 3.4 are as follows:

MODE : The propagation mode having the greatest reliability.

FIGURE 3.2 EXAMPLE PAP OUTPUT

(COURTESY OF PROF. M. DARNELL, UNIVERSITY OF HULL)

```
(HFMUFES 4 77/4/30)
JUNE 21,     LOW        10 CM FLUX  71          (SSN 10)
H5 SVL(SHIP TX)  TO  SITE 105(RX)      AZIMUTHS       MILES    KM.
50.00N - 20.00W      69.20N - 16.00E  28.73 240.46  1778.5 2862.1
                     MINIMUM ANGLE 2.0 DEGREES
POWER=.25KW    3MHZ NOISE=-165.0DBW  REQ.REL.=.85  REQ.S/N=50.0DB
                MUF(****) - FOT(++++) - LUF(....)
```

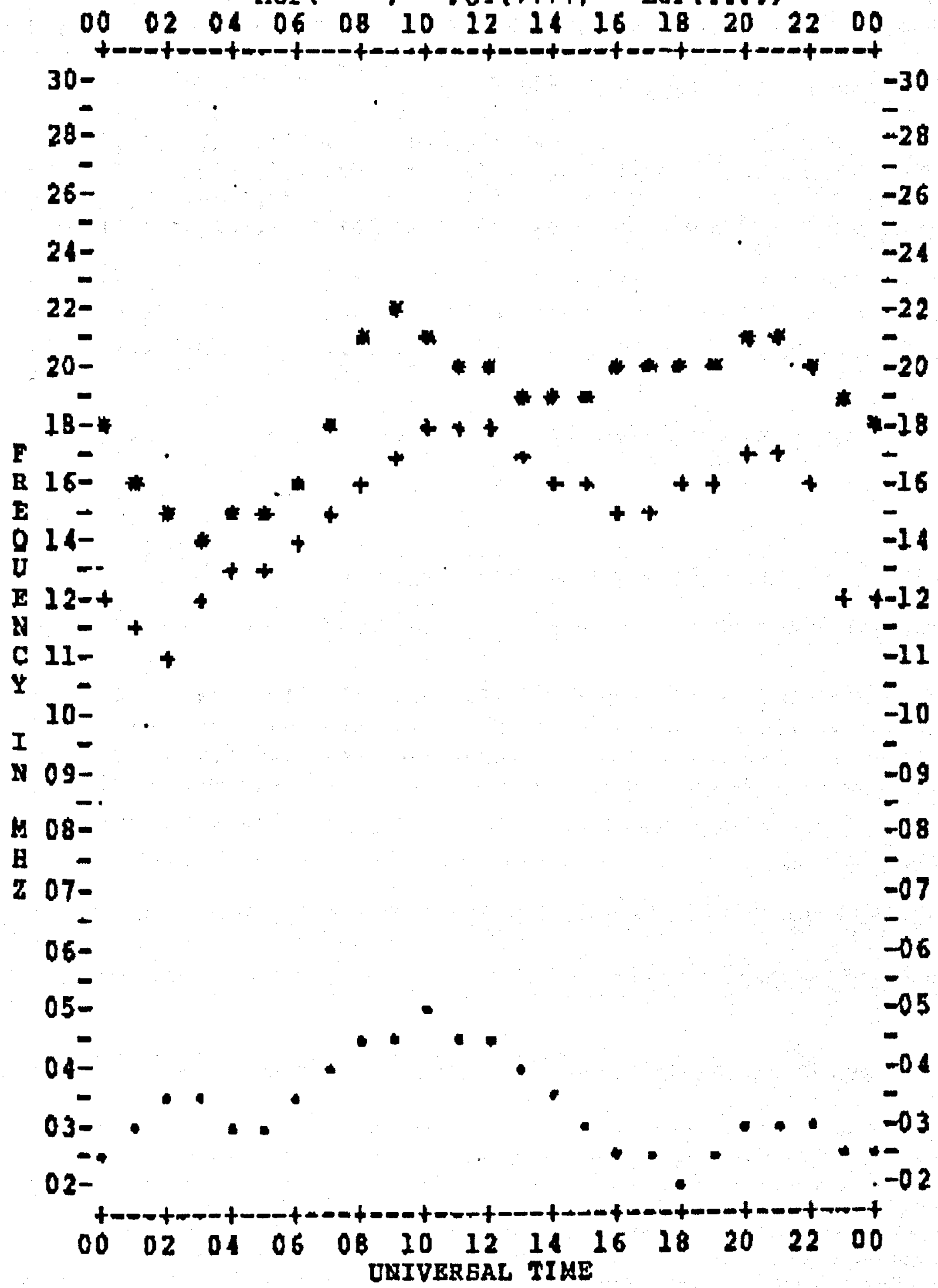


FIGURE 3.3 EXAMPLE PAP OUTPUT

(COURTESY OF PROF. M. DARNELL, UNIVERSITY OF HULL)

```
                                          15                 (HFMUFES 4 77/4/301
                  JUNE 21,  MED    10 CM FLUX 102       TSSN  301
G5 SVL(SHIP TX) TO SITE 506(PX)                     AZIMUTHS      MILES      KM.
50.00N - 20.00W  49.45N -  7.55E  -  111.83  314.36    1304.1    2098.7
                              MINIMUM ANGLE  2.0 DEGREES
XMTR  2.0 TO 30.0     VERTICAL     H   0.00 L   -.25 A   0.0 OFF AZ    0.0
RCVR  2.0 TO 30.0  CONSTANT GAIN   H   8.00 L   0.00 A   0.0 OFF AZ    0.0
POWER=    .25KW   .3 MHZ NOISE=-165.0DBW   REQ. REL=.85 - REQ.S/N=50.0DB
                                  RELIABILITIES
```

UT	MUF	2.1	3.2	5.0	6.9	8.0	11.4	12.1	14.6	16.1	19.0	25.0	MUF
01	12.8	.52	.58	.62	.66	.67	.55	.47	.14	-	-	-	.51
02	11.7	.50	.52	.59	.62	.63	.43	.33	-	-	-	-	.39
03	11.2	.34	.51	.61	.66	.68	.34	.21	-	-	-	-	.38
04	11.3	.05	.48	.56	.66	.69	.36	.23	-	-	-	-	.38
05	11.8	-	.29	.57	.69	.72	.49	.35	.04	-	-	-	.40
06	12.6	-	-	.56	.64	.72	.61	.50	.11	-	-	-	.41
07	13.5	-	-	.55	.63	.68	.70	.63	.24	-	-	-	.41
08	14.4	-	-	.26	.56	.61	.69	.66	.40	.12	-	-	.39
09	15.4	-	-	.15	.52	.59	.70	.68	.49	.21	-	-	.77
10	16.4	-	-	.07	.48	.57	.55	.68	.50	.23	-	-	.77
11	17.1	-	-	-	.46	.58	.51	.73	.51	.72	-	-	.84
12	17.5	-	-	-	.43	.56	.47	.76	.47	.19	-	-	.83
13	17.6	-	-	-	.42	.55	.46	.74	.43	.16	-	-	.78
14	17.6	-	-	-	.43	.55	.44	.73	.40	.14	-	-	.78
15	17.1	-	-	-	.39	.51	.70	.46	.37	.11	-	-	.76
16	16.1	-	-	-	.37	.47	.66	.64	.36	.10	-	-	.79
17	14.6	-	-	-	.24	.33	.57	.58	.30	.14	-	-	.35
18	15.3	-	-	.31	.26	.34	.59	.61	.32	.26	-	-	.38
19	16.2	-	-	.16	.42	.48	.63	.65	.60	.43	.06	-	.41
20	16.7	-	.07	.26	.46	.52	.64	.65	.62	.47	.10	-	.39
21	16.4	-	.18	.40	.50	.55	.67	.67	.61	.44	.07	-	.40
22	15.4	.21	.30	.46	.54	.58	.67	.67	.53	.20	-	-	.40
23	14.4	.42	.53	.59	.64	.67	.65	.62	.41	.18	-	-	.42
24	13.4	.47	.56	.60	.65	.67	.63	.36	.28	.06	-	-	.42

FIGURE 3.4 EXAMPLE PAP OUTPUT

(COURTESY OF PROF. M. DARNELL, UNIVERSITY OF HULL)

PATH=4000KM TX POWER=1.0KW REQ.S/N=60DB SSN=20

GMT MUF OPERATING FREQUENCIES

GMT	MUF	3	4	5	6	7	8	10	12	
06	8.3	2F2	2F2	2F2	2F2	2F2	2F2	2F2		MODE
		7	5	5	5	5	6	7		ANGLE
		187	185	185	185	186	187	187		DELAY
		50	99	99	94	81	57	10		C.PROB
		51	52	55	53	55	55	60		S/N D/B
		50	28	76	89	80	57	10		REL
08	7.3	2F2	2F2	2F2	2F2	2F2	2F2			MODE
		7	5	5	5	6	7			ANGLE
		187	185	185	185	186	187			DELAY
		50	99	97	85	59	27			C.PROB
		52	49	54	54	53	61			S/N D/B
		48	29	71	76	56	26			REL

ANGLE	:	The median vertical angle associated with the above propagation mode.
DELAY	:	The propagation delay time in 0.1 ms units for that mode.
C.PROB	:	The percentage of days for which the mode is expected to exist. This is also known as the circuit probability.
S/N DB	:	The monthly median SNR for the mode with the greatest available received power, when that mode is present. SNR is defined as (total signal power in transmission bandwidth ÷ noise power in a 1 Hz bandwidth).
REL	:	The product of the availability of that mode with the greatest availability and the probability that the mode with the greatest received power, when present, provides a monthly median SNR exceeding that required.
MUF	:	Highest frequency with an availability of 50%.
FOT	:	Sometimes called the optimum working frequency (OWF). It is the highest frequency with an availability of 90%.
LUF	:	The lowest frequency for which the monthly median SNR for the mode with the greatest received power, when present, exceeds the SNR specified for the proportion of time specified by the required reliability. LUF may also be defined more simply as the lowest frequency giving the required reliability.

In common with HFMUFES4, most PAPs have a wide range of output formats available.

3.3.3 Available Types of PAP

There are many HF analysis programs with varying levels of complexity. A brief outline of the main types of PAP now follows (Thrane, 1983):

(i) The CCIR recommended method based upon CCIR report 252-2 (CCIR, 1970). These routines require mainframe computer support.

(ii) Methods based upon the supplement to CCIR report 252-2 (CCIR, 1978). Mainframe support is required.

(iii) IONCAP (Lloyd et al, 1981). This program makes SNR forecasts over the frequency range of 2 to 55 MHz. It also has a model of sporadic-E mode propagation superior to that contained in (i) and (ii). The original version of this program runs on a mainframe, although there is now a pc-based version available (pc-IONCAP).

(iv) APPLAB. This is a mainframe-based program which was developed at The Rutherford Appleton Laboratories, UK (Bradley, 1975).

(v) Minicomputer-based prograns such as, MUFLUF, FTZ (Damboldt, 1980), LIL 252 and HFM YLE.

(vi) Microcomputer based programs such as MINIMUF-3 (Levine et al, 1978), and its successor MINIMUF-3.5 (Rose, 1982), the procedures detailed in (Devereux & Wilkinson, 1983) and (Gerdes, 1984), MICROP-2 (Dick & Miller, 1987) and MICROP-3 (Dick & Lewis, 1989).

As can be seen from (vi) above, considerable research effort has been spent in recent years on the development of microcomputer-based PAP's. MICROP-2 was developed at Rutherford Appleton Laboratories, UK and it makes use of the compact prediction methods detailed in CCIR report no. 894 (CCIR, 1986). It comes complete with a "user-friendly" input routine and graphical output displays. The object code resides in 181 kbytes of RAM and the ionospheric data necessary to produce predictions is held in a database of size 280 kbytes. It has six output options and when run on an IBM pc AT it requires approximately 50 seconds of execution time to produce output data for a 24 hour period. "Cut-down" versions of the methods presented in CCIR report no. 252-2 (CCIR, 1978) were previously used to provide microcomputer-based predictions and these take approximately three minutes to produce equivalent output data. The accuracy of MICROP-2 is comparable with prediction methods based upon CCIR report no. 252-2.

MICROP-3 is an updated version of MICROP-2 having an improved user-interface and more output options.

3.4 Shortcomings of PAPs

The common characteristics of propagation prediction routines have already been described. PAPs of all sizes have some common disadvantages associated with their use and these are detailed below:

(i) The databases of ionospheric measurements used to generate PAPs are limited for certain regions of the earth's surface, such as the large oceans and the polar zones.

(ii) The physical understanding of ionospheric propagation mechanisms is limited in some respects. This can lead to uncertainties in the resulting predictions made with PAPs.

(iii) The short-term, ie minute-by-minute changes occuring on an HF path (which is of great interest to HF communicators) cannot be adequately predicted using algorithms designed to produce long-term, median-type outputs.

(iv) There is no input of information concerning co-channel interference.

Items (i) and (ii) above will become less significant as more HF propagation data is gathered and more accurate propagation models are developed.

Item (iii) above can be minimised by the injection of ionospheric short-term forecasting data. The principal source of short-term variability is in the value of the vertical incidence critical frequency for the F_2 layer, f_oF_2 (Rush et al, 1974). If real-time data concerning f_oF_2 is injected into the PAP (via measurements made by a vertical incidence sounder) then the effects of this variability could be reduced. The use of RTCE within the frequency management system of an HF communication system will also provide information about the short-term behaviour of a path.

Black-outs (due to sudden solar flares etc) and MUF perturbations are other sources of channel variability. If the pre-cursors of these conditions can be detected, then it may be possible to re-route communications traffic (assuming a network scenario, with stations covering a large geographical area) to avoid the disturbed control points.

Co-channel interference, ie that originating from other communications signals, has been shown to be a serious problem in spectrally congested regions of the earth. Indeed, in areas such as Western Europe, it can limit the performance of an HF communications system. There is thus a need to predict co-channel interference conditions on HF channels if reliable communications are to be achieved.

3.5 Co-Channel Interference Modelling

The types of noise and interference associated with HF radio channels are as follows:

(i) Receiver noise (usually assumed to be thermal).

(ii) Galactic noise.

(iii) Atmospheric noise, eg that from thunderstorms.

(iv) Man-made noise, eg from electrical machinery, arc-welding equipment etc.

(v) Harmonics and inter-modulation products due to non-linearities in the RF equipment.

(vi) Co-channel transmissions from other users of the band.

Noise sources (i) - (v) can be thought of as broad-band effects, being fairly predictable across the whole HF band whereas noise source (vi) is generally narrow-band.

As detailed in Chapter 2 (Section 2.3), receiver noise is insignificant in the HF band, when compared with man-made and atmospheric noise. Noise sources (ii), (iii), and (iv) are modelled by PAPs, as described in Sections 3.3.1 and 3.3.2. The generation of harmonics and intermodulation products by any RF radiating system can be minimised by careful design of the RF stages and the appropriate treatment of metal antenna support structures. As yet, there is no adequate means of modelling HF co-channel interference. Some quantitive models have been produced for specific locations, examples of which are given below.

Figure 3.5 shows the model described in (Wilkinson, 1982). It was found that the measured cumulative probability of a co-channel signal being above a specific level can be described by a normal distribution function, for a range of bandwidths.

Figure 3.6 shows an occupancy model within a 3 kHz channel bandwidth (Darnell, 1979). For measurement purposes, the 3 kHz channel has been divided into 16 sub-channels. The occupancy characteristics fit a negative exponential function. The probability that n sub channels (not necessarily contiguous) of bandwidth δf are occupied by a signal or signals with level(s) greater than a threshold L_T is:

$$p(n) = k_1.\exp(-k_2.n) \tag{3.1}$$

$$1 \leq n \leq 15$$

where k_1 and k_2 are constants for a given value of L_T.

The peak which occurs at n = 16 is not included in the model. It is thought to be due to speech and medium-speed data signals filling the complete 3 kHz channel.

In order to produce a general co-channel interference model, there is a need for a comprehensive, co-ordinated data collection programme. One such programme has

FIGURE 3.5. GENERAL FORM OF CUMULATIVE DISTRIBUTION FOR CO-CHANNEL SIGNAL LEVELS

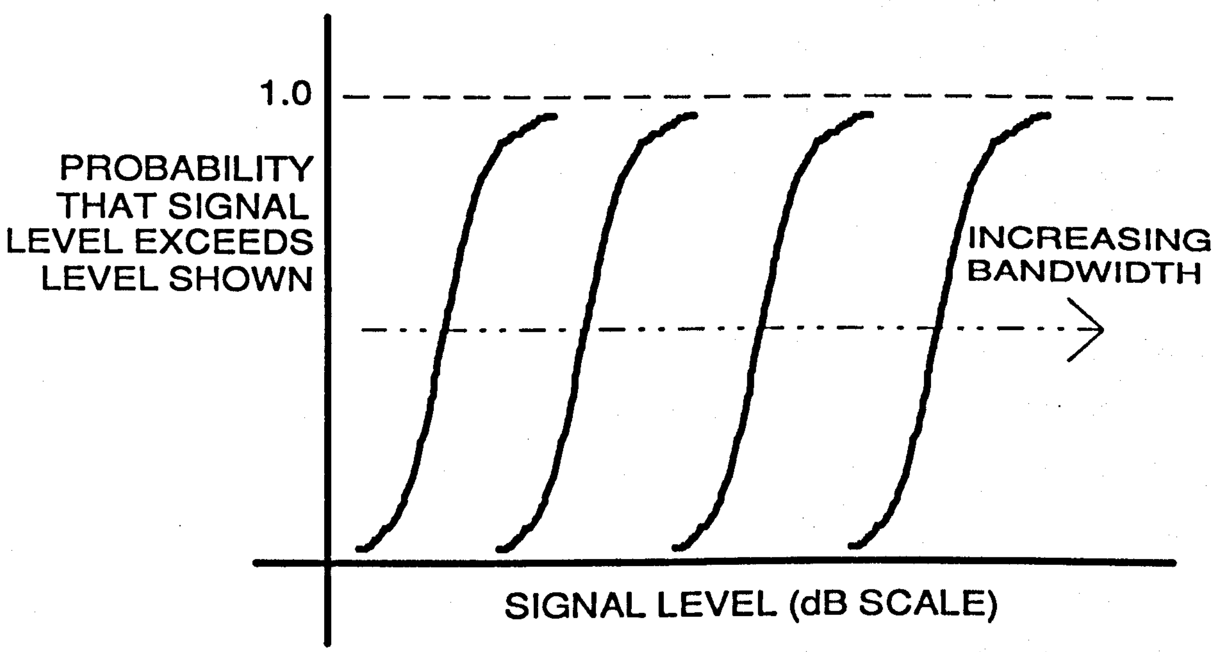
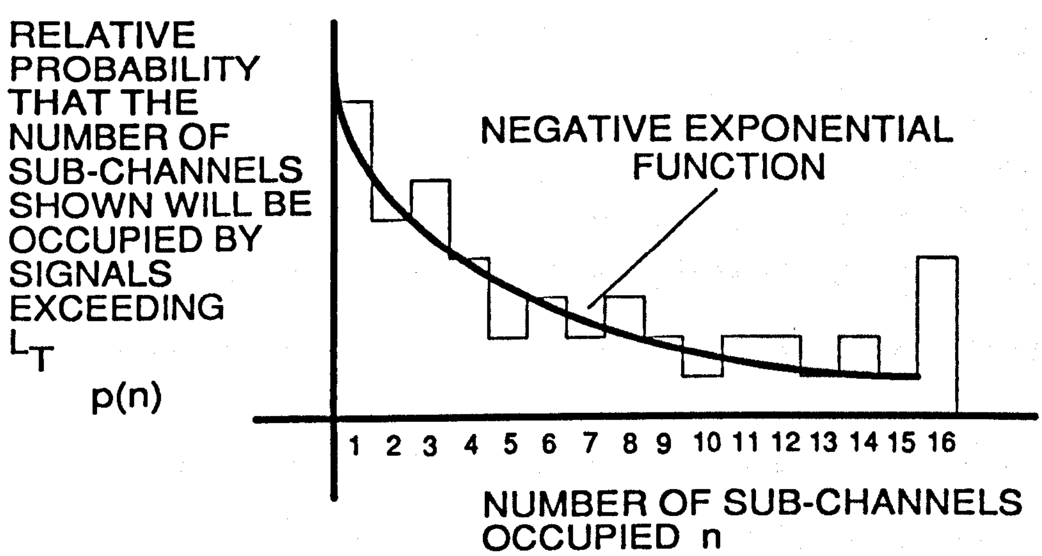


FIGURE 3.6. MODEL FOR OCCUPANCY OF 3 kHz BANDWIDTH CHANNEL



been under way for just over half a sun-spot cycle (5-6 years) and it is attempting to characterise the UK interference environment (Gott et al, 1982). The main aims of this work are to produce an overall spectrum occupancy model and to yield data for a more detailed and accurate statistical model.

The basis for the resulting statistical model, described in (Laycock et al, 1988), is the set of measurements made at the winter and summer solstices. For the purposes of each measurement campaign, the HF spectrum was analysed in terms of a congestion value, Q, at five threshold levels, for 95 defined frequency sub-bands, where Q is defined as the probability of finding, at random, a 100 Hz bandwidth, within a 50 kHz spectrum where the average interference level exceeds a defined threshold.

The resulting database consists of approximately 3 million measurements made using a measurement bandwidth of 1 kHz, stepped through 1 kHz increments in order to achieve approximate de-correlation between adjacent values of Q (earlier work by Gott et al had shown values of Q to be independent for spacings of 1kHz or more).

Generally Q will be a function of several parameters:

$$Q = f(a,b,c....) \qquad (3.2)$$

where a,b, and c may be sun-spot number, location frequency etc.

A form of the model which has been shown to fit the data well is:

$$Q = \exp(y) / (1 + \exp(y)) \qquad (3.3)$$

where $0 \leq Q \leq 1$ and $-\infty < y < \infty$.

y is given by equation 3.4 below :

$$y = A_i + B_o P_{th} + (C_o + C_1 f_i + C_2 f_i^2).S \tag{3.4}$$

where:

A_i : has 95 values corresponding to the 95 sub-channels for which the interference measurements were taken.

P_{th} : threshold value in dBm.

B_o : multiplying coefficient.

C_0, C_1, C_2 : coefficients of a quadratic equation in f_i where f_i is the centre frequency of the i^{th} sub-band where $1 \leq i \leq 95$.

S : sun-spot number.

Equation 3.3 is only applicable to a 1 kHz bandwidth.

This model is currently being refined further to take account of interference in sub-channels corrresponding to specific user groups, eg fixed, mobile, aeromobile, maritime mobile, broadcast and amateur. These user groups have previously been shown to exhibit very different interference characteristics (Gott et al 1982). Also, other bandwidths and the effects of geographical location are being examined.

From the work already performed on characterising HF co-channel interference, the following observations can be made:

(i) Interference is predominantly narrow-band in nature, ie with a bandwidth less than 3 kHz (Darnell, 1979).

(ii) The average interference profiles in a 3 kHz channel persist for several minutes before significant changes occur (Gott and Stanniforth, 1978).

(iii) In Central/Western Europe, the probability of finding a 2.5 - 3 kHz channel clear of significant interference energy is low, especially at night (Gott et al, 1982).

Over the past two decades, off-line propagation analysis has been increasingly complemented by various forms of RTCE - the subject of the following chapter.

4 EMBEDDED RTCE TECHNIQUES

4.1 Introduction

RTCE is a pre-requisite for efficient and effective frequency management of automatic HF communication systems. RTCE routines provide short-term information about the overall behaviour of HF channels and thus they can be used to overcome some of the problems associated with embedded HF PAP's.

A generalised treatment of RTCE is given in (Darnell, 1983). The majority of the original RTCE techniques require equipment additional to that ordinarily present at an HF terminal which is used for communications purposes. This requirement inhibited the widespread use of RTCE because of economics: the cost of the RTCE hardware was, in many cases, comparable with that used for communications purposes.

This chapter concentrates on embedded RTCE techniques, ie those which use the existing communications terminal hardware to provide path parameter measurements. They are thus less expensive to implement, from a hardware point of view, than stand-alone RTCE routines. The major disadvantage of such routines is that they have to be incorporated into the terminal at the design stage. However, this can be overcome to a certain extent by the use of structured system architectures and design methodologies, such as those detailed in Chapters 9 and 10.

Long-term PAP's were reviewed in Chapter 3 and they have the following shortcomings:

(i) The databases of ionospheric measurements used to generate the PAP's are limited for certain regions of the earth, eg near the poles and over large oceans.

(ii) The physical understanding of ionospheric propagation mechanisms is limited in some respects which leads to uncertainties in the resulting predictions.

(iii) The short-term variability of HF paths cannot be adequately described using algorithms that were designed to produce long-term, median-type output.

(iv) The effects of co-channel interference on circuit performance are not included in PAP's.

RTCE can help to overcome shortcomings (iii) and (iv) above. (Items (i) and (ii) will become less significant as more ionospheric data is gathered and thus better models of path behaviour are produced). RTCE routines do not provide predictions of future path behaviour: they evaluate a system's frequency allocations at the current time. Hence, a number of HF FMS's use a combination of PAP's and RTCE routines to meet both long and short-term path data requirements.

4.2 A Review of RTCE

The definition of RTCE adopted by the CCIR is as follows (CCIR, 1981):

> "Real-time channel evaluation is the term used to describe the processes of measuring appropriate parameters of a set of communication channels in real-time and of employing the data thus obtained to describe quantitively the states of those channels and hence their relative capabilities for passing a given class, or classes, of communication traffic".

The following points about the above definition should be noted (Darnell, 1986):

(i) RTCE is primarily concerned with deriving a numerical model for each of the system's frequency allocations in a form which can be used for short-term performance prediction and system control purposes.

(ii) The channel model generated by RTCE should be appropriate to the type of traffic transmission being employed, eg the model generated for a low-rate telegraphy link should be different from that used for a digitised speech link.

(iii) The term "real-time" means that the measured channel parameter values are updated at time intervals which are less than the overall response time of the communications system to control inputs. Updating too often would mean that the resulting information could not be used effectively and it would thus be redundant.

(iv) It is desirable that the output from an RTCE routine should be expressed in a form which is meaningful to the communicator/system controller, eg a predicted BER for digital transmissions.

(v) RTCE is concerned with the characterisation of the propagation, natural noise and co-channel interference conditions experienced on a channel. It can thus compensate for some of the shortcomings of PAP's.

(vi) In addition to providing optimum transmission frequency information, RTCE should also provide data concerning the optimum start-times and durations of transmissions.

(vii) RTCE techniques should also be capable of selecting channels which can make use of transient propagation modes, eg sporadic E-layer refraction, in order to increase the availability of a link.

Figure 4.1 shows a generalised RTCE algorithm with the basic four inputs used by most routines. Many different forms of RTCE routine have been developed. Measurable path and channel parameters which have been, or could be, used by RTCE algorithms include (Darnell, 1986):

(i) Signal amplitude.

(ii) Signal frequency.

(iii) Signal phase (absolute or differential).

(iv) Propagation time (absolute or relative).

(v) Noise or noise and interference level.

(vi) Channel impulse response function.

(vii) Signal-to-noise or signal-to-interference ratio.

(viii) Energy distribution within the channel bandwidth.

(ix) Received digital data error rate.

(x) Received speech intelligibility level.

(xi) Telegraph distortion factor.

(xii) Rate of repeat requests in an ARQ system.

4.3 RTCE Algorithms

In the context of an automatic HF communication system, RTCE routines serve two purposes:

(i) To evaluate the current channel that is passing the communications traffic.

FIGURE 4.1 A GENERALISED RTCE ALGORITHM

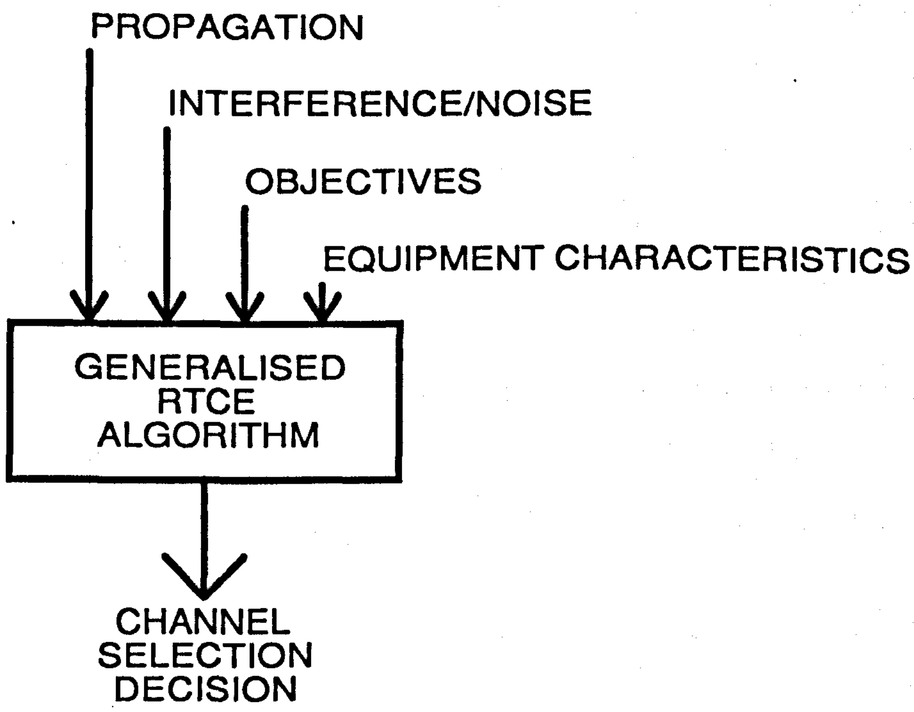
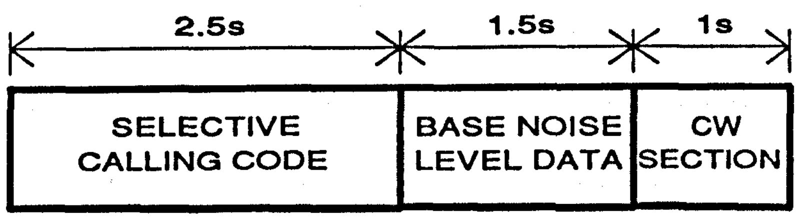


FIGURE 4.2 CHEC SIGNAL FORMAT



(ii) To evaluate the alternative frequency allocations, in order to allow a ranked list of channels to be constructed.

There are two general types of RTCE routine:

(i) PASSIVE RTCE, ie that which derives its channel assessment input from the normal communications traffic or "transmissions of opportunity".

(ii) ACTIVE RTCE, ie that which employs active probing of channels to produce RTCE output.

Several RTCE algorithms are presented below, all of which have been constructed and tested to at least a prototype stage. The techniques described in Sections 4.3.1, 4.3.2, and 4.3.3 are all classed as active RTCE techniques, the remainder being passive.

4.3.1 Channel Evaluation and Calling (CHEC) System

CHEC was developed in Canada to improve air-to-ground communications for maritime patrol aircraft (Stevens, 1968). It was designed to aid communications from a number of mobile stations to a single base station.

On each of m assigned channels, where m is usually less than 20, the CHEC base station radiates in sequences a probing signal which is of the format shown by Figure 4.2. The relevant base station is alerted by the reception of its own calling code. When this occurs, the base station average noise levels, given by:

$$\left.\overline{n(t)}\right|_{f_i} \qquad 1 < i < m \tag{4.1}$$

are decoded for the k channels that actually propagate to the mobile station to give:

$$\left.\overline{n(t)}\right|_{f_j} \qquad 1 < j < m \tag{4.2}$$

where j can take any k distinct values in the range 1 to m and also $k < m$.

The subset of corresponding average received signal levels at the mobile are:

$$\left.\overline{A(t)}\right|_{f_j} \tag{4.3}$$

These are evaluated using the CW sections of the base transmissions .

By assuming propagation reciprocity and by making what allowances are possible for the differences in the transmitter powers and the antenna configurations at the base and mobile stations, a processor at the mobile computes the average SNR for transmissions from the mobile to the base in each of the k channels, as follows:

$$\left. \mathrm{SNR}(t) \text{ base} \right|_{f_j} = \left. \left[\frac{G.\overline{A(t)}}{\overline{n(t)}} \right] \right|_{f_j} \tag{4.4}$$

where G is a channel dependent factor to compensate for the differences in antenna and transmitter characteristics between base and mobile stations.

The optimum channel, as predicted by the CHEC system, is given by the value of j for which the SNR is a maximum.

CHEC was shown to give significant improvements in channel availability and reliability during trials carried out with it in a prototype form. Other systems that are similar in concept to CHEC are described in (Wynne, 1979) and (Chow et al, 1981). A more modern system, known as ALQA, is described in (Bliss, 1985). It forms the basis of the LQA automatic HF communication system which is discussed in Chapter 5.

4.3.2 RTCE by Pilot Tone Phase Error Measurement

In this scheme an RTCE probing signal, in the form of a CW tone, is inserted at a suitable position within the current channel bandwidth (Betts and Darnell, 1975). Phase variations in this tone are measured at the receiver and the channel quality is estimated by making use of analytical relationships between phase instability and BER.

The experimental system compared the phase of the received pilot tone with that of a locally-generated reference phase source. The phase difference was sampled at regular time intervals and the phase difference of sample n was compared with that of sample n-1. Ideally, the difference between the two sample phases should be zero: however, for practical HF channels it will be non-zero. When the difference in the measured phase between the channels exceeds a certain threshold value, θ_t, then a "phase error" is counted, ie

$$\text{when } |\theta_n - \theta_{n-1}| > \theta_t \qquad (4.5)$$

then a phase error occurs.

The sampling interval needs to be an integral multiple of the pilot tone period so that sampling occurs at the same point in the tone cycle, under ideal conditions. The parameter used to describe the state of a channel is the number of phase errors occuring in a pre-determined interval (usually 100 - 200 s).

In the experimental trials a low-level pilot-tone was frequency multiplexed with a 50 bits/s two-tone FEK signal, the pilot-tone being inserted at a null in-between the FEK tones. The system was calibrated by calculating the number of phase and bit errors over the same measurement period, for particular channel conditions.

The experimental trials results showed that, for the majority of channel conditions encountered, the data error rate which would be experienced could be predicted with reasonable accuracy.

This RTCE scheme could be employed in a multiple mobile, single base station scenario, similar to that of the CHEC system described in Section 4.3.1. The base station would emit the CW tone and, by assuming propagation reciprocity and allowing for differences in base and mobile station RF equipment and antennas, a channel error rate prediction for mobile to base transmissions could be made. The long channel evaluation time required by this scheme is offset by the simplicity and low-cost of the implementation at the mobile stations.

4.3.3 RTCE by Error-Counting

RTCE by error-counting evaluates HF channels by means of a test signal having the same format as the traffic signal. The most convenient version of this RTCE technique uses digital, binary data transmissions to evaluate the channel so that the channel state can be quantified directly. Speech intelligibility could be used for channel analysis purposes but this tends to be subjective and more difficult to quantify. A degree of synchronisation is necessary between the transmitter and receiver, in both time and frequency, but it does not need to be as accurate as that required for ionospheric sounding methods, such as chirpsounding (see Chapter

7).

A known probe sequence is sent over all the system's channels and the number of bit errors occurring in the probe is counted at the receiver. This error count is a direct and traffic-related measure of a channel's acceptability, since the signal format used by the RTCE routine is the same as that used for communications purposes.

Practical trials of this technique are detailed in (Darnell, 1978). 75 bits/s FEK telegraphy and 1200 bits/s digital data signals were transmitted over 700 and 1100 km paths and the trials were carried out 24 hours/day. A PAP was used to produce a two-frequency operating schedule of the type shown by Figure 3.1, and trials were carried out using this schedule to control the link. RTCE was then used to provide frequency management information for the same period of time and the results from the two sets of trials were compared. It was found that the circuit availability was increased by approximately 45% in the RTCE-controlled trials. This was thought to be due to the fact that the interference levels experienced during the trials were relatively high. RTCE enabled the communications system to avoid interference related problems rather than just tracking the propagation changes on the path.

RTCE by error-counting can be used in speech transmission systems. In this case, a digital probe is used as the basis for channel evaulation. The error-count at the receiver is related to speech intelligibility from calibration data, thus enabling the channel quality to be assessed.

4.3.4 Chirpsounding

RTCE information can be gained via passive monitoring of chirpsounder transmissions. Chirpsounders are a type of oblique ionospheric sounder (Barry, 1966). Chirpsounder transmitters send a CW tone which is swept linearly in frequency, with respect to time, across the whole HF band (2 to 30 MHz). In order to receive and demodulate the chirp signal, chirpsounder receivers possess a time and phase synchronised replica of the transmitted waveform. This is used, in conjunction with the incoming chirp waveform, to derive the impulse response of the path from the chirpsounder to the monitoring site, for the whole band, in the form of an ionogram.

There are 44 known chirpsounder transmitters, spread fairly evenly around the world. These can be monitored passively by HF communication system terminals and the path parameters derived can be used to enhance the performance of PAP's. Full details of the chirpsounding ionospheric measurement system, corresponding PAP enhancement techniques, and a relatively inexpensive chirpsounder reception system are given in Chapter 7.

4.3.5 In-Band RTCE

This is concerned with the evaluation of sub-channels within a 3 kHz bandwidth. Techniques include measuring the energy within the sub-bands, enabling signals to be placed away from high levels of interference and noise. A deterministic technique for achieving this within the context of an automatic HF communications system is known as template correlation (Jowett and Darnell, 1987). Full details of this are presented in Chapter 8.

Alternative methods include maintaining a running assessment of SNR in an FSK/FEK system. It is possible to monitor the noise/interference level within a given tone sub-channel, when the other tone is keyed on. This level is used, along with the signal tone power to compute the SNR at the receiver. The running SNR assessment can be made for both the current and alternative channels (Humphrey and Shearman, 1985).

4.3.6 RTCE using Soft Decision Information

Soft decision information can be extracted from the received signal amplitude, or the phase margin between a phase reference and the phase of signals detected by a receiver. Generally, any information that can be used to quantify a detection decision can, in principle, be used for RTCE purposes.

In a DPSK modem, eg KINEPLEX (Mosier and Clabaugh, 1958) the phase margin between a received signal, $\theta_r(t)$, and a locally generated reference phase, $\theta_o(t)$, could be used as the basis of a channel assessment scheme. In this case:

$$\text{RTCE assessment} = f\{\ |\theta_r(t) - \theta_o(t)|\ \} \qquad (4.6)$$

Alternatively, the RTCE assessment could be a function of the amplitude of the received signal, as in the CODEM system (Chase, 1973).

4.3.7 RTCE in ARQ Systems

The number of block repeat-requests in a given time interval can be used as a measure of the channel quality. The channel with the lowest block repeat-request rate is thus the highest in quality.

FIGURE 4.3 VARIABLE REDUNDANCY CODING IN AN ARQ SYSTEM

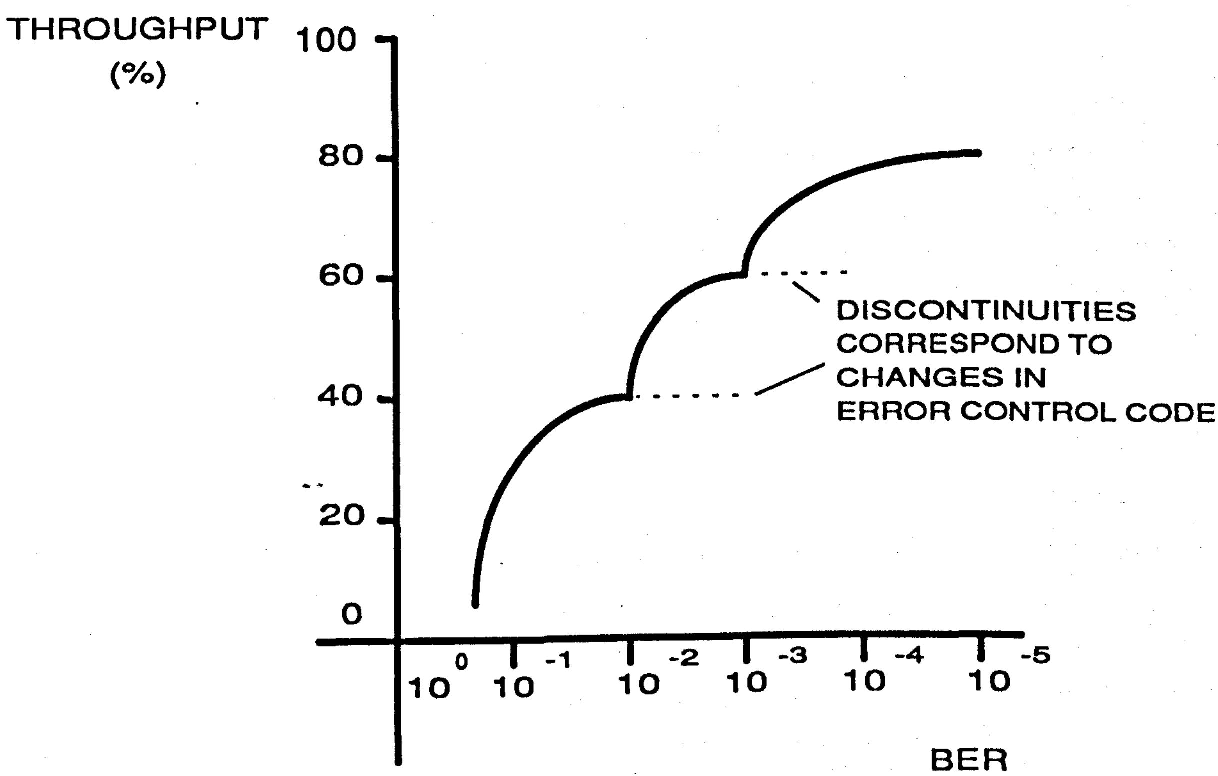
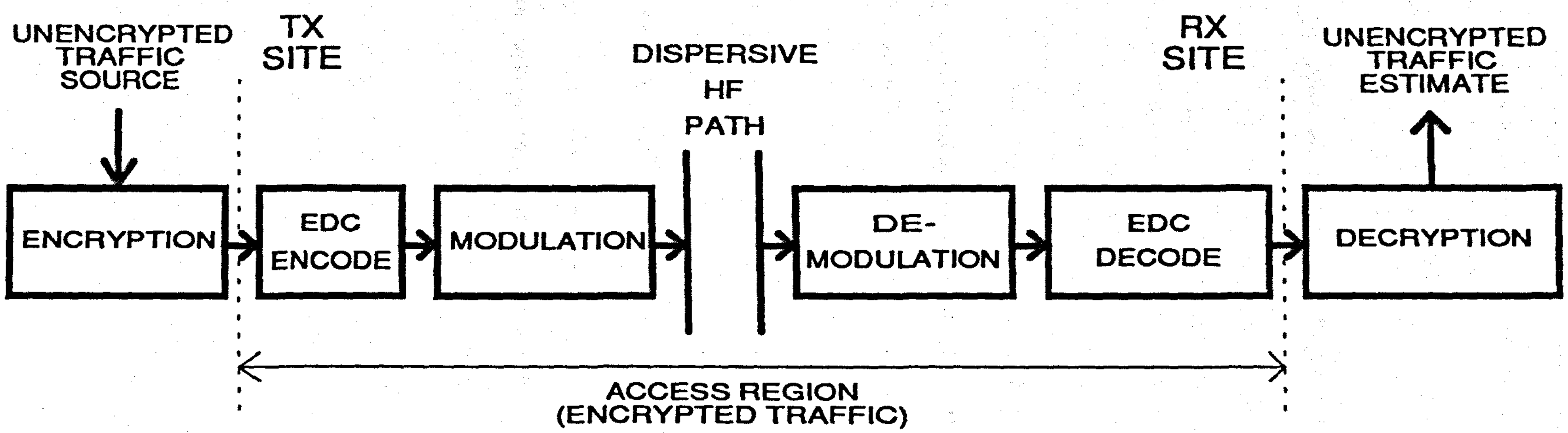


FIGURE 4.4 THE USE OF AN AUXILIARY EDC SYSTEM FOR RTCE WITH ENCRYPTED DATA



With fixed-length data blocks, encoded using an error-control code, it is possible to vary the power of the code used according to the block repreat-request rate. This is known as "variable redundancy coding" (Goodman and Farrell, 1975). As the BER of the forward path increases then new codes are selected at specific BER thresholds. Thus the throughput of the system is adjusted to match the prevailing channel conditions whilst maintaining a minimum BER at the output of the ARQ system. Figure 4.3 shows typical characteristics for such a system.

4.3.8 RTCE by Traffic Signal Modification

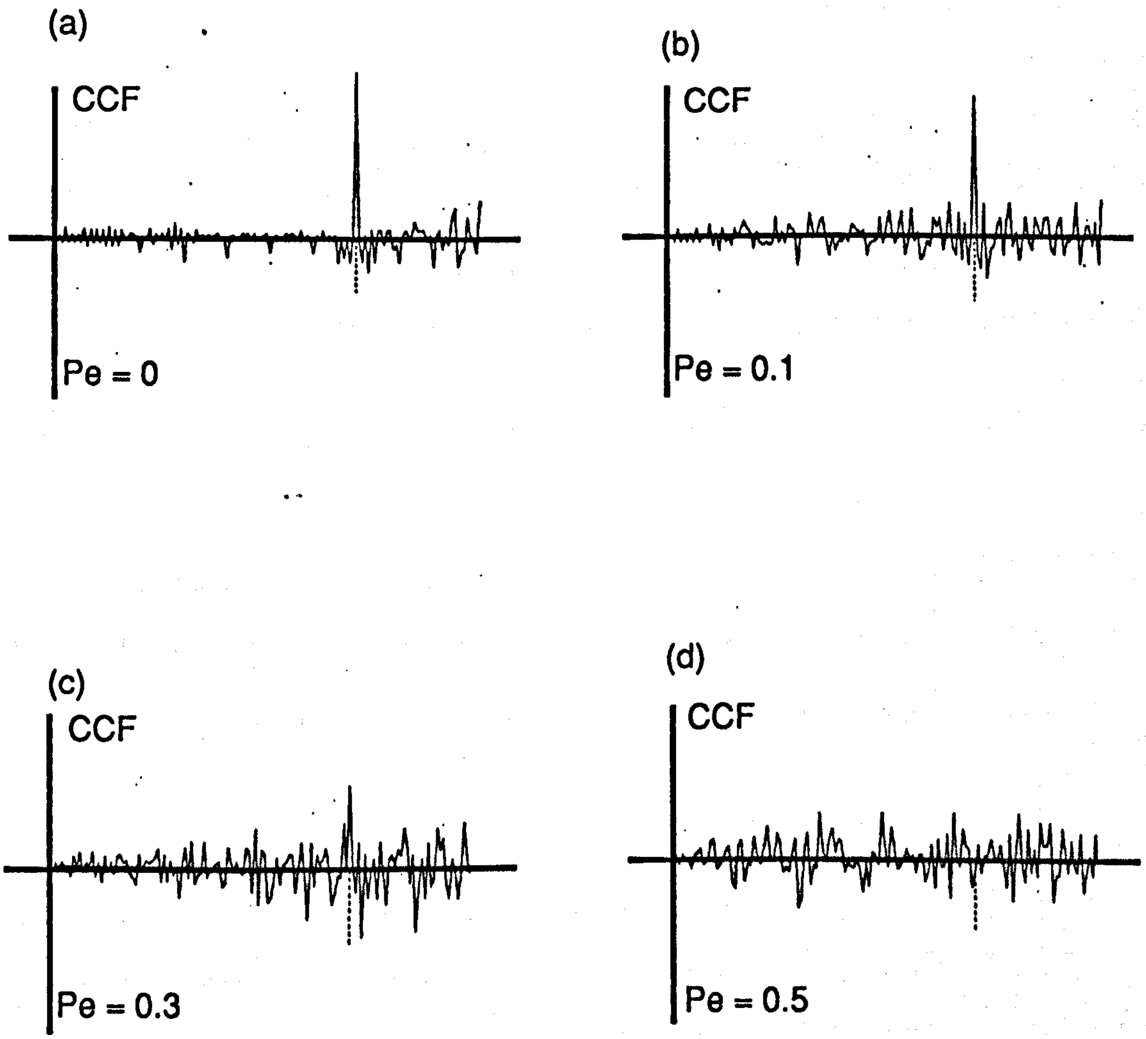
This technique is used when it is not possible to obtain any RTCE information from the traffic, due to security restrictions or there being no soft-decision information available. A low-level pilot tone can be inserted at a suitable point in the signal spectrum (as described in Section 4.3.2). Alternatively, specialised error control techniques can be employed when access to the plain text data is unavailable due to security restrictions, or because the data is encrypted. In this scenario, an auxiliary EDC scheme is used, as shown by Figure 4.4. The encrypted traffic is formatted into arbitrary codewords and the received codeword error rate is used to assess channel quality.

4.3.9 RTCE by Assessment of Synchronisation Quality

Synchronisation pre amble correlation results can be used to assess channel quality. Figures 4.5(a)-(d) show the performance of a synchronisation pre amble, comprising of a 77 bit code produced from concatenated 7 and 11 bit Barker sequences (Barker, 1953), in Gaussian white noise. The traces show the output of a detection filter (cross-correlator) matched to the synchronisation sequence, for the error rates shown. It is seen from these that the peak-to-sidelobe ratio of the synchronisation

FIGURE 4.5. CONCATENATED BARKER SEQUENCE CORRELATION FUNCTIONS IN GAUSSIAN WHITE NOISE

DOTTED LINE INDICATES POSITION OF CORRELATION PEAK
Pe = PROBABILITY OF ERROR (DUE TO NOISE)
CCF = CROSS-CORRELATION FUNCTION



(Courtesy of Dr. E.D. Chesmore, University of Hull) results decreases with increasing channel BER; this ratio could be used to assess channel quality.

4.3.10 RTCE by Pseudo-Error Counting

This is used to overcome the long analysis times associated with error-rate monitoring of low BER channels (Leon, 1973). The error-count measurement, detailed in Section 4.3.3, is amplified by making use of an over-sensitive detection scheme so that the error-rate measured is substantially greater than that experienced by the communications traffic. Thus the error count for low BER channels will accumulate more rapidly than with conventional error-count assessment algorithms.

It may be difficult to apply this technique to HF links due to the inherent high error-rates associated with HF channels. Also, the rapid time variation in signal attributes would lead to difficulties in calibration.

Short intervals of a modulation technique, requiring a higher SNR than the actual traffic signal for successful reception, could be inserted into the traffic, eg short bursts of 1200 bits/s DPSK inserted into a stream of 75 bits/s FEK telegraphy.

It is also possible to use "soft error" schemes. If the average demodulator output during a symbol interval is within a certain amplitude band then a soft error is registered. Widening the soft error band will thus increase the sensitivity of the system.

4.4 Potential Advantages of RTCE for HF Communicators

The following is a summary of the advantages that HF communicators may experience through use of RTCE techniques (Darnell, 1986):

(i) The dependency on PAPs, which have inherent short-term inaccuracies, will be reduced. A combination of RTCE and PAPs can produce efficent frequency management schemes (see Chapter 5).

(ii) The effects of man-made interference on system performance can be measured and quantified, thus enabling the error-free capacity of a particular communications link to be enhanced.

(iii) Relatively transient propagation, eg sporadic E-layer refraction can be identified and used for communications. This can enhance the availability of a particular link.

(iv) RTCE enables the selection of channels higher in frequency (when available) than would be suggested by PAP output alone. This serves to reduce spectral congestion and increase link availability.

(v) RTCE provides a means of automatic optimum channel selection and facilitates the ranking of alternative, stand-by channels - a pre-requisite for effective automatic HF system frequency management.

(vi) Radiated power levels and spectral occupancy can be minimised as the communications traffic signals can be adjusted to meet specific signal fidelity requirements. This serves to reduce spectral pollution.

(vii) Data can be extracted from RTCE routines to aid the adaptation of the communications system parameters, eg signal processing techniques, error control methods, antenna characteristics etc.

In general, RTCE techniques are to provide current path performance data for automatic HF communication systems. Current HF communication systems use them, in conjunction with PAP's to provide operational frequency management information (see Chapter 5 for a review of current automatic HF communication system architectures). In a typical HF FMS, PAP's are used to provide the predicted channel performance data, on which channel selection decisions are made, before a communications link is established. Thus, due to the inherent inaccuracies of PAP's, a system may well be operating initially using a non-optimum channel. RTCE routines are used after a link has been established. They measure the performance of current and alternative channels, thus enabling the system to home in on the optimum channel for the particular transmission requirements.

RTCE is thus a pre-requisite for effective HF frequency management. Because of their ability to provide current path performance parameters, RTCE techniques will continue to play an important role in future HF FMS's, such as those detailed in Chapters 9 and 10.

5 OVERVIEW OF THE PLESSEY AUTOMATIC HF COMMUNICATION SYSTEM

5.1 Rationale for Automating HF Communication Systems

Prior to the proliferation of computers and other binary data sources, eg vocoders, HF radio communication systems were used primarily to carry analogue speech and morse transmissions over long-haul paths. Such systems were controlled manually by skilled operators. The operator had to decide which out of a number of channels was optimum, using a propagation-derived frequency operating schedule, and his experience of operating on a particular link, to provide the necessary frequency management information. The system would possess limited or no adaptivity, eg fixed transmitter power and transmission mode (USB, LSB etc).

Three main factors are leading to a progressive automation of HF communication systems:

(i) A change in the nature of HF communications traffic.

(ii) The inadequacies of manual control regimes.

(iii) A decrease in the price/performance ratio of computing equipment.

As stated previously, analogue speech and morse code messages originally constituted a large proportion of the traffic passed over HF paths. Due to the increasing use of digital computers, electronic facsimile machines, and digital encoding schemes for speech, binary data is now the main source of HF communications traffic.

It is necessary to automate the link management and flow control algorithms for digital data commmunications. With human operators and analogue speech as the traffic, human processing power at each end of the link can be used to overcome the effects of burst errors on a channel. A knowledge of the context and meaning of the message is employed in order to achieve this. However, this would not be possible given the same error conditions on a link passing binary data. The data might be encrypted and it could originate from any one of a number of potential sources, eg facsimile machines, computers, vocoders etc. Thus a human operator would be unable to compensate for such errors effectively. Error control algorithms and ARQ schemes are necessary to ensure "wire-like" data transmission.

Additionally, automated systems allow easy interfacing to the major sources of HF traffic. Also, if the system is software-controlled, it is potentially easy to alter its configuration, thus enabling the system to adapt to changes in operating environment.

In manually controlled HF communications systems control data is typically supplied by off-line, long-term PAP's. These are used to produce a simple frequency operating schedule, often requiring the operator to change the working frequency as little as two times per day. Natural noise and interference data for a particular channel would come from the operator's experience gained through use of the system over a particular path. As mentioned above the system would have limited or no adaptivity.

Such control regimes are inefficient for digital data transmissions, due to the nature of a typical HF path. In Chapter 2 it was seen that the channel capacity of a particular path varies greatly within a short time interval. A non-adaptive system would not be able to react to, or exploit, such variations in channel capacity. Periods of high channel capacity would thus be wasted. Human operators are prone to errors and they also have a relatively slow response to changes in path conditions. This again leads to channel capacity wastage.

The shortcomings of PAP's were presented in Chapter 3. A system will not be operating effectively if it relies upon the frequency management data supplied by HF PAP's alone. There is also limited input concerning levels of co-channel interference. It has been shown that co-channel interference can be the factor limiting HF communication system performance in spectrally congested, mid-latitude regions.

It is only relatively recently that it has become economically feasible to embed significant levels of processing power within the terminals of HF communication systems. The processing power previously associated with minicomputer systems is now available within a desk-top pc package. These factors enable PAP's and other large prediction algorithms to be incorporated into each terminal of an automatic HF communications system. Special-purpose signal processing systems such as the TMS320C25 can also be incorporated with the architecture of proprietary pc systems.

Generally, the problems associated with manually controlled HF communication systems can be alleviated by automating system control and by incorporating adaptive system components into the terminal configuration. The overall response time to changes in path conditions should decrease. Also, automation facilitates the use of embedded PAP's and RTCE routines, which together provide long-term path trends and short-term path behaviour statistics. Incorporating embedded prediction and channel analysis routines and adaptive equipment into the architecture of a terminal allows the system to function in a more flexible manner and it enables frequency management decisions to be made more rapidly. Computers, if programmed correctly, are more reliable and accurate than humans when performing repetitive and defined analytical tasks. Automation also allows the logging of link performance parameters so that trends in path usage and interference statistics can be derived. This type of output is useful when revising the configuration of an HF terminal. Automation provides the speed of response and degree of adaptivity required for successful HF data communications.

5.2 The Plessey Automatic HF System

The basic features of this system (which forms the framework for much of the research work described in this thesis) are as follows:

- Fully automatic system control.
- Embedded propagation and noise models.
- Embedded RTCE via error-counting.
- Noise/interference assessment via passive monitoring of frequency allocations.
- FEK modulation with the option of in-band diversity or multiplexed operation.
- Data rates of 75, 150 and 300 bits/s.
- Link control via ARQ scheme.

The terminal hardware used is shown in Figure 5.1. The frequency management routines and overall system control algorithms are resident on the IBM pc XT. The two BBC Model B microcomputers are responsible for link synchronisation at both block and bit-level. The modems used are Cossor CGT 1092 units. These have switched-capacitor filtering in the demodulator and they are fully-programmable via circuit board-mounted switches. The transceiver is fully computer-controllable via an RS-232 interface. A broadband, portable HF antenna (Andrews type 4065) is used which facilitates transmitting over frequency range 2 to 30 MHz, whilst maintaining a VSWR of no greater than 2:1.

The control software for the system is divided into separate program modules which are loaded in to the pc from hard disk as required. The software is written in

FIGURE 5.1. PLESSEY SYSTEM HARDWARE CONFIGURATION

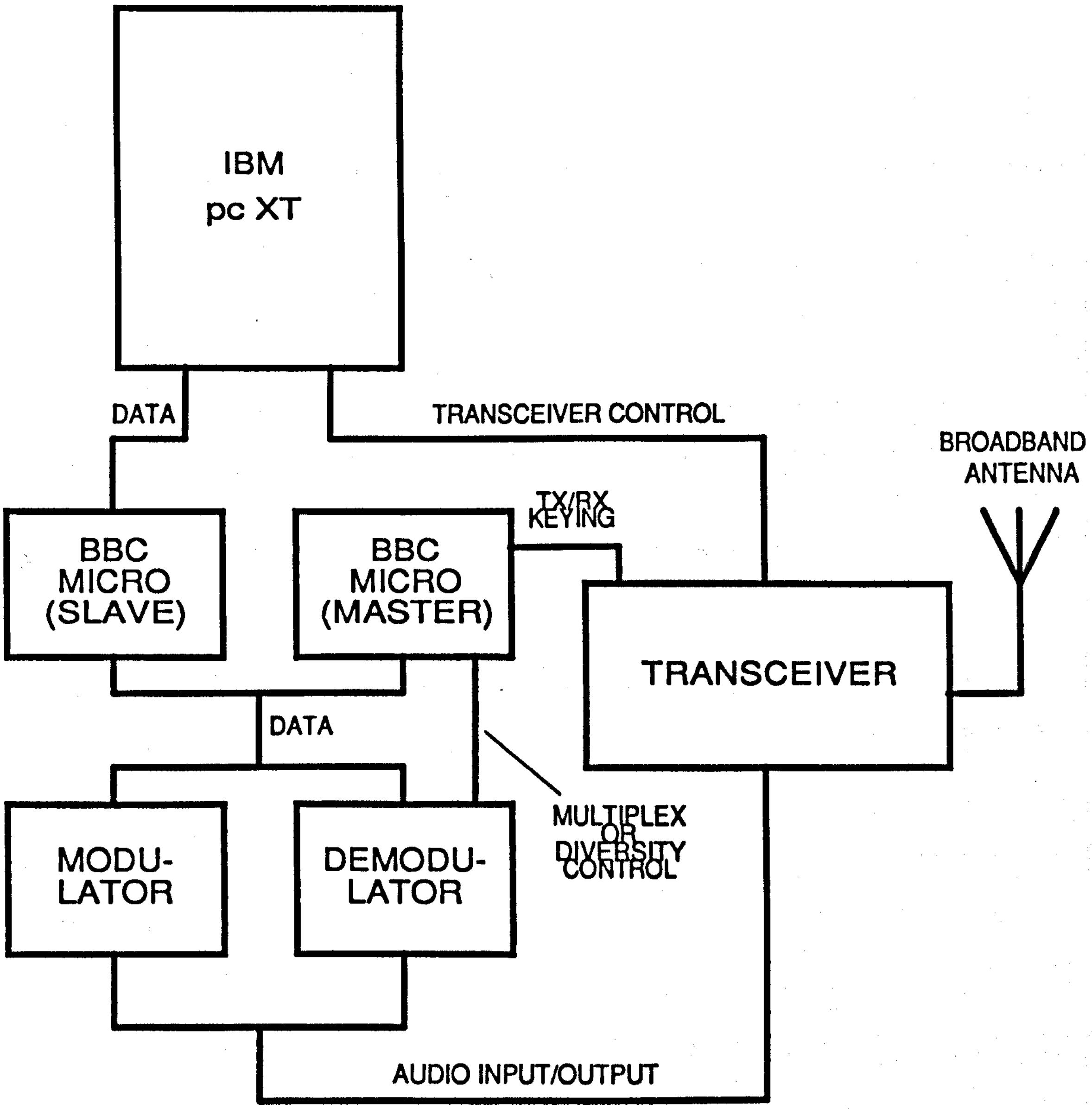


FIGURE 5.2. PLESSEY SYSTEM FMS ALGORITHM

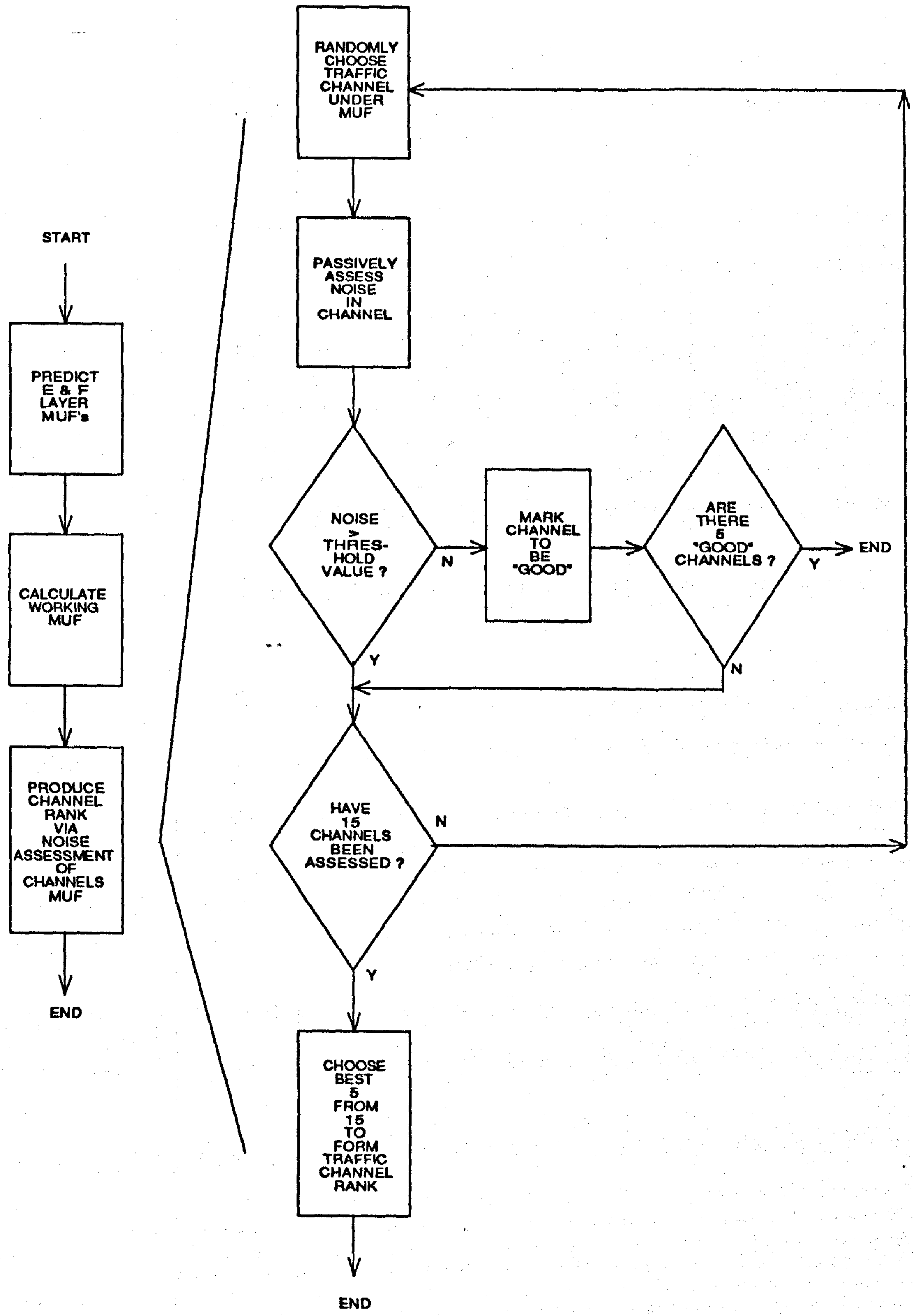


FIGURE 5.3
(a). SETUP ALGORITHM: SOURCE

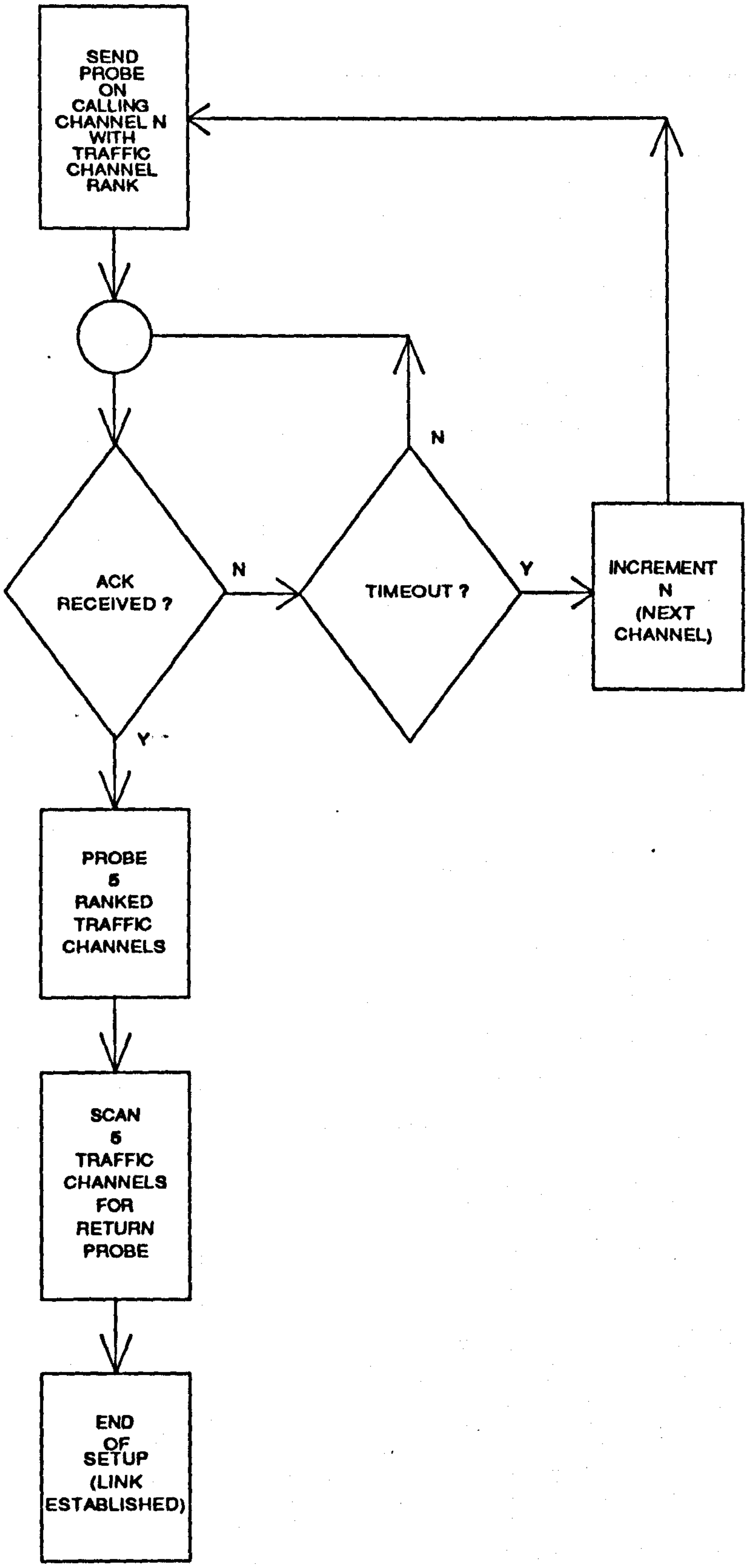
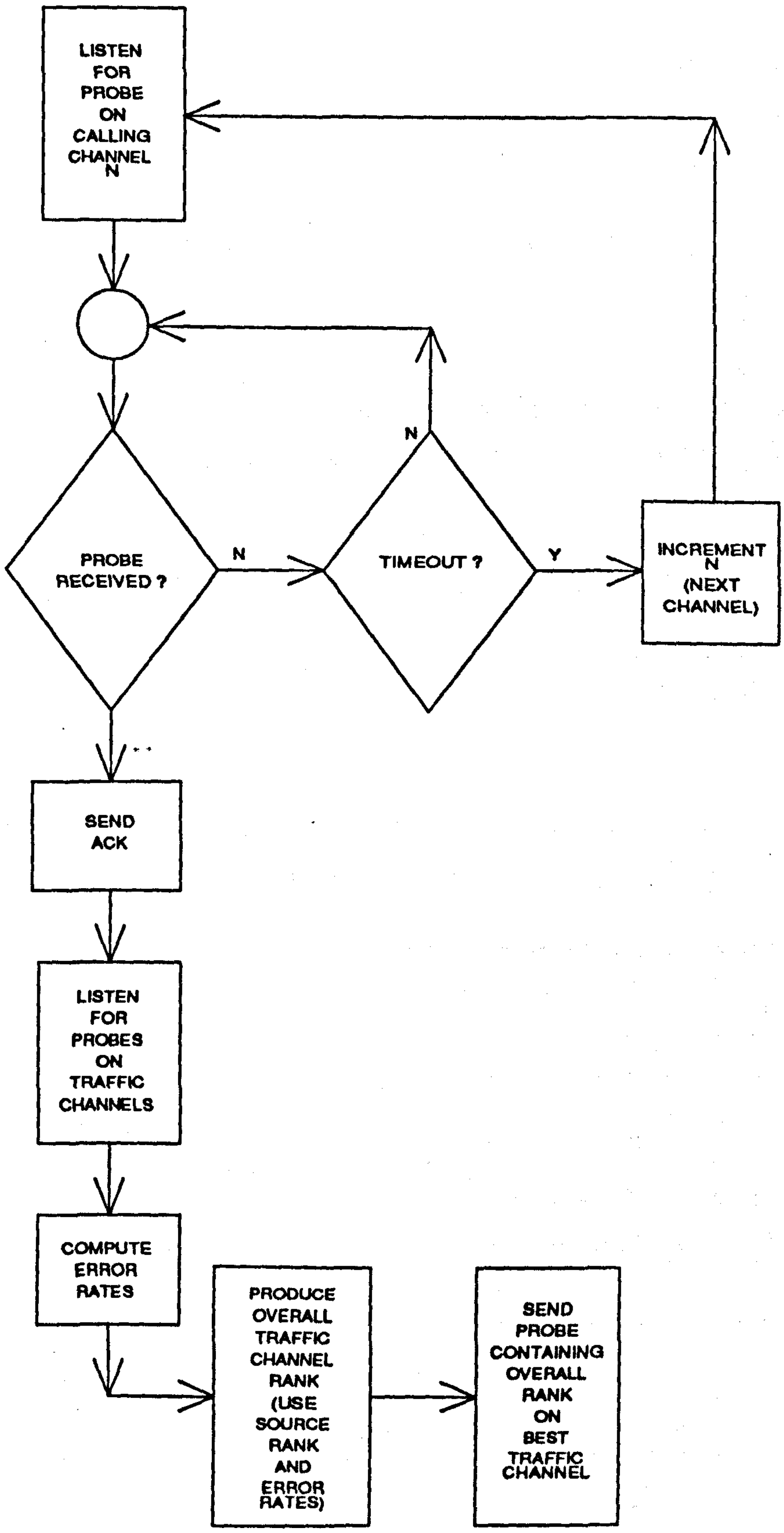


FIGURE 5.3
(b). SETUP ALGORITHM : DESTINATION



compiled BASIC, Pascal and 6502 Assembly Language. A short description of each of the software modules now follows:

MENU This is the user interface to the system. It facilitates message entry and system configuration alterations. It also selects the mode of operation of the system, eg manual, automatic-non-adaptive (fixed data rate), automatic-adaptive (with the ability to adjust the data rate according to path conditions).

FMS FMS (frequency management system) contains the prediction model for the system. The algorithm used by the FMS is shown in Figure 5.2. It predicts E and F-layer MUFs, using the last measured MUF from the RTCE routines and the predicted MUF to produce the so-called "working" MUF via an averaging procedure. Passive noise assessment of the channels which propagate is then carried out according to the algorithm shown in Figure 5.2.

SETUP This software module is responsible for the initialisation of calls for a particular path. The algorithm used by the setup routines is shown in Figure 5.3. SETUP ensures that end-to-end synchronisation is achieved and it ensures that each end of the link has the correct traffic channel ranking. SETUP also contains the RTCE-by-error-counting module, BANDSCAN. This probes the system's traffic channels by sending a known binary sequence over the path which is then analysed for errors at the receiver. A BER figure is then produced for the system's frequency allocations. Odd and even-numbered channels are probed at half-hourly intervals (this schedule can be adjusted via the user i/o routines).

TX & RX These modules handle the transmission and reception of data, and error and link flow control. Error control is provided by means of a Golay (23,12) encoding-decoding scheme and link flow control is effected by use of a selective ARQ algorithm.

In addition to the software modules listed above, there are assembly language routines resident on the two BBC microcomputers. These routines provide bit-synchronisation and re-timing of data and are resident in EPROMs in the machines. Also there are a number of data files resident on the main processor containing information such as frequency allocations, station data etc.

Figure 5.4 shows the operational cycle used during the development trials of the system.

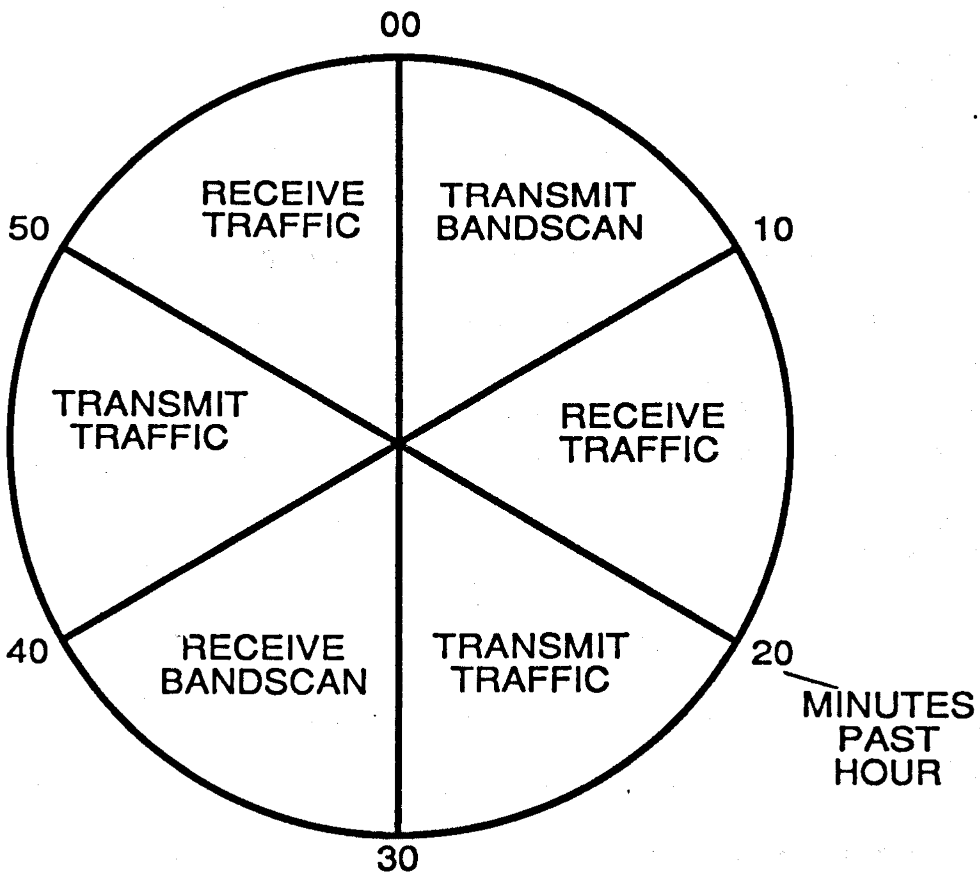
In order to aid further development of this system a compatible terminal was constructed at the University of Hull. The hardware used at the Hull site differed from the hardware at the Plessey sites: the Hull terminal incorporated an amateur-grade transceiver and an IBM pc XT clone instead of the professional-grade transceiver and an IBM computer used in the Plessey terminals. Considerable time was spent trying to integrate this new hardware into the existing terminal architecture.

Modification of the transceiver was necessary so that received signal strength readings could be extracted from it. This was achieved by constructing an ADC board, attaching the analogue input to the signal strength meter on the transceiver and feeding its digital output to one of the pc's i/o ports. Interface software was written to convert the port reading into a signal strength percentage value.

The serial control interface on the amateur-grade transceiver operated via signals using TTL voltage levels. A circuit was constructed which converts these TTL voltages to RS-232 compatible levels, thus allowing the transceiver to be

FIGURE 5.4. OPERATIONAL CYCLE FOR TRIALS OF THE PLESSEY SYSTEM

(COURTESY OF PLESSEY RESEARCH LTD.)



SCHEDULE IS FOR ROKE MANOR END OF TRIALS LINK.
TO GET TIMINGS FOR OTHER END, SWAP SEGMENTS ABOUT THE CENTRE.

automatically controlled by the pc.

The original control software was modified as the control protocols for the Hull transceiver differed from that of the unit used at the Plessey sites. Also timing differences were experienced between the IBM pc and the pc clone used at Hull. The IBM pc XT uses the 8-bit Intel 8088 microprocessor. The pc clone at the Hull site uses a 16-bit Intel 8086 microprocessor. Both machines have the same clock frequency. Memory accesses on the IBM require 3 machine cycles whereas on the clone, only two cycles are used because of its 16-bit architecture. Hence the clone appears to run approximately 1.5 times faster than the IBM machine for the same application. This caused problems with software timing during back-to-back trials and considerable effort was spent trying to eradicate this.

A compatible broadband antenna was installed on a roof-top site at Hull. This was a portable field antenna of the same type used at the Plessey sites. Modification to the antenna structure was necessary in order to facilitate roof-top mounting.

In the initial stages of the research programme, considerable time was spent in defining and formalising the existing Plessey software elements and their interactions. This was necessary to allow a full understanding of the system operating and also to facilitate future software/protocol development.

5.3 Other Contemporary Automatic HF Communication Systems

There are many such systems in existence including the following:

(i) ARTRAC, developed by SRT AB, Sweden.

(ii) ALQA-SELSCAN, developed by Rockwell-Collins, USA.

(iii) CQS-based systems, developed by the Andrews' Corporation, Australia.

(iv) ASSATS, developed by GEC-Marconi Ltd, UK.

A brief description of systems (i)-(iii) now follows for comparison with the Plessey system (the Author was unable to find detailed information pertaining to (iv)).

5.3.1 ARTRAC

ARTRAC (automatic radio traffic controller) was developed by SRT AB, Sweden to provide single channel, low-rate data and analogue voice communications (Sjoebergh, 1988). The system is fully automatic with the provision for manual control for testing purposes. Frequency management information is supplied during the link set-up phase by an embedded propagation prediction model and passive assessment of the noise and interference levels in the system's frequency allocations. In-band RTCE, via assessment of the noise levels in the FSK demodulator filters, provides the system with in-call frequency management data. Link flow control is provided via an ARQ scheme and the data rate is fixed at 125 bits per second.

The system is currently undergoing further development.

5.3.2 ALQA-SELSCAN

ALQA and SELSCAN are registered trademarks of the Rockwell-Collins Corporation. ALQA is an abbreviation for automatic link quality analyser and SELSCAN is the name given to the routines used to establish an HF communications link. ALQA and SELSCAN have been used, together with a supervisory microprocessing unit, to form the basis of an HF communications system (Bliss, 1985). The processor unit was based on an Intel 8086 device with an 8087 maths co-processor, ie it had similar processing power to an IBM pc XT. This unit is responsible for remote control of a transceiver, for the execution of an error control scheme and for supervision of the SELSCAN and ALQA modules. The system uses FSK modulation at either 75 or 300 bits/s.

The ALQA module provides RTCE information for the current and alternative channels. Fast Fourier transforms are performed on the channels and the results of these are used to derive values for the signal to noise power spectral density ratio, the fading rate and depth, the fading power spectrum, the mean and RMS frequency offset, the noise and interference frequency, time and amplitude statistics, and figures for the harmonic distortion experienced over the path.

The SELSCAN routines are responsible for the setting up and intialising of a communications link. Use is made of an embedded PAP and passive noise assessment of the system's frequency allocations.

5.3.3 CQS

CQS is an abbreviation of channel quality sounder and it was developed by the Andrews' Corporation of Australia. It is an RTCE scheme which has been used in the trials of a low-rate telegraphy system (George & Halligan, 1985).

A PAP is used before a call is in progress to derive a list of usable frequencies. CQS is then used to decide which channels are usable and which one is optimum at a given time. CQS derives channel parameters by making measurements on CW tones transmitted over the required path.

Unfortunately, no information was available regarding the performance of this technique.

5.4 Inadequacies of the Plessey Automatic HF Communications Systems

It is seen from the above descriptions that the Plessey system is similar in features to the above systems. Unfortunately, performance data was unavailable for the various systems which would have allowed more meaningful comparisons to be made. However, the inadequacies of the Plessey system were identified as being the following:

(i) Channel selections are made on the basis of data provided by PAP's, which have been shown to be inaccurate when used to provide short-term data (see Chapter 3 for a discussion of the problems associated with these programs).

(ii) There is limited input concerning levels and types of co-channel interference and an inability of present systems to adapt signal formats to avoid interferers.

(iii) The systems have been designed without any formal design methodology and little structure. This impedes the maintenance of the system and makes it difficult to incorporate new system components into a terminal and test them.

The eradication of the above inadequacies was the main objective of this research programme, as is detailed in the following chapter.

6 AIMS OF THE RESEARCH PROGRAMME

6.1 Enhancement of HF PAP Performance

Chapter 3 showed current propagation prediction algorthms to be inadequate for use in automatic HF communication systems. Their short-term accuracy is poor, which reduces the effectiveness of an embedded frequency management regime which uses these programs. The poor, short-term perfomance of such programs can be compensated for, to a limited extent, by embedding RTCE routines into the system. However, certain HF users are required to transmit for the minimum possible period of time, and when transmitting, to radiate the least amount of power. Operating in this way minimises the probability of the geographical location of the terminal being determined by an eavesdropper. These users are unable to employ active RTCE, as this would increase the time spent transmitting and thus it would increase the chance of detection. Additionally, the amount of information that can be extracted from passive RTCE routines would be limited, due to the restrictions imposed on the transmission of traffic. These so-called "quiet" users must rely, for their frequency management information on HF PAP's.

Improving the accuracy of HF PAP's would benefit the "quiet" users and it would increase the efficiency of all automatic HF communication systems. In particular the Plessey HF system would become more efficient as a more accurate propagation model should enable it to identify an optimum channel more often, before communications traffic is passed over a link.

6.2 Extraction of Interference Information

It was stated in Chapter 3 that an adequate model of HF co-channel interference was currently un-available. In the stage of operation before traffic is passed over an HF link, ie when the frequency management routines are trying to predict which out of a number of channels is likely to be optimum, interference information is normally acquired by passively monitoring the frequency allocations which propagate for a given path. This does not provide channel usage trend information, nor does it provide the system with information concerning the optimum position for placing a signal within the communications bandwidth. Due to the importance of adequate interference input for automatic HF communication systems operating in spectrally congested regions it was decided to investigate ways in which the above could be achieved.

6.3 Automatic HF Communication System Design

Chapter 5 reviewed the Plessey HF system and some other state-of-the-art automatic HF communication systems. There was no co-ordinated methodology used in the design of the Plessey system. A structured, logical approach to the design of such systems would allow configuration changes to be made relatively easily. The system could be easily adapted to changes in operating environment. Also, a degree of compatibility between HF and other long-range communication systems would give communication station controllers the ability to re-route traffic as required, eg when conditions are poor, or when terminals in a network are taken out of service for maintenance etc.

It was thus decided to develop a design methodology for an automatic HF communication system.

7 PROPAGATION MODELS AND METHODS OF ENHANCING THEIR ACCURACY

7.1 The Need For Accurate Propagation Models

In a typical automatic HF communications system, propagation models are used to provide pre-call path parameter information, such as the MUF and the path reliability (see the description of operation of the Plessey HF System in Chapter 5). It is the aim of the pre-call phase to decide which, out of all the user's frequency allocations, is most likely to be optimum. Propagation models thus play a major role in the making of this decision.

The discussion in Chapter 3 shows that the accuracy of propagation models is good in the long-term, but poor on an hour-to-hour or minute-to-minute basis. As these relatively short-term characteristics of the desired path are required in order to make a channel selection decision, then it is probable that a propagation model will provide the frequency management system, embedded within the overall communications system, with erroneous information. In the worst case, a channel may be labelled as having the ability to propagate when it is unable to do so.

The most common way of minimising the effects of these inaccuracies is to incorporate RTCE into the frequency management system. In this way the model is used initially to provide an approximate indication of the likely conditions on a channel and the system's parameters are fine-tuned as a result of the RTCE output. At first the system may be operating inefficiently, but it will rapidly "home in" on the optimum channel, once the communications link has been established.

Results from trials of the HF communications system developed by Plessey Research Ltd. have shown that the average throughput rate is highest when the system initially sets up on the optimum channel. An accurate propagation prediction model would ensure that the optimum channel was initially found more often and it would thus increase the average throughput rate.

There are certain categories of user, eg the military, who do not wish to give away the locations of their HF terminals (especially those who are mobile). In order to try and avoid detection they must transmit using the minimum required power and reduce the duration of transmission to a minimum. They may also wish to ensure that their communications traffic does not propagate to certain geographical areas. Active RTCE, ie that which derives path parameters from the transmission and reception of probing signals (Darnell, 1986), is undesirable in this scenario and the information available from passive RTCE (the source of which is the communications traffic itself) is minimised, due to the transmission requirements. These factors severely limit the ability of the frequency management system to select the optimum channel during the in-call phase of the communication system's operation. A propagation model with enhanced accuracy would potentially increase the average throughput rate of the system and would also reduce the probability of detection.

Methods of increasing propagation prediction model accuracy are examined below.

7.2 Enhancing Propagation Model Accuracy

Techniques for improving model accuracy generally have three phases in their operation : a measurement phase, a comparison phase and an adjustment phase. Initially a propagation parameter is measured, such as the MUF for a particular path. The propagation model is then run using input parameters relevant to the same path and its output value for the selected parameter is then compared with the measurement. The model is then adjusted via, for example, input parameter manipulation, to make the measurement and prediction agree and this adjustment is then used in all subsequent predictions until the next update occurs.

An example of propagation model accuracy enhancement following this mode of operation is given in (Uffelman et al , 1984). The method used was as follows: the MUF for the path between a receiving station and an oblique-incidence ionospheric sounder was measured and recorded over a 24 hour period. A prediction model was used to produce a 24 hour MUF characteristic for the same path and the two curves were then compared. A best fit was achieved by altering the value of the input sun-spot number to the model. This so called "pseudo sun-spot number" was used in all subsequent propagation prediction made during the next 24 hours, when the process was again repeated.

The accuracy of the prediction model was increased using this method. In some cases the degree of improvement was such that the error between the enhanced prediction and the actual MUF was equal to the minimum error achievable, as estimated by the designers of the model.

Other methods of increasing model accuracy have been examined. Figure 7.1 shows how the data from two measurement sources could be used to interpolate propagation information about a third path. In order to investigate the validity of this technique, the third, estimated path could be that to another propagation measurement system , such as a sounder, thus allowing comparison between interpolated results and actual path performance to be made.

Figure 7.2 shows how the ionosphere could be probed 1 hour to the East of the required refraction, or control, point (Rush at al, 1974). Information from this measurement would then be valid approximately one hour later. This is in effect

FIGURE 7.1. PROPAGATION DATA INTERPOLATION

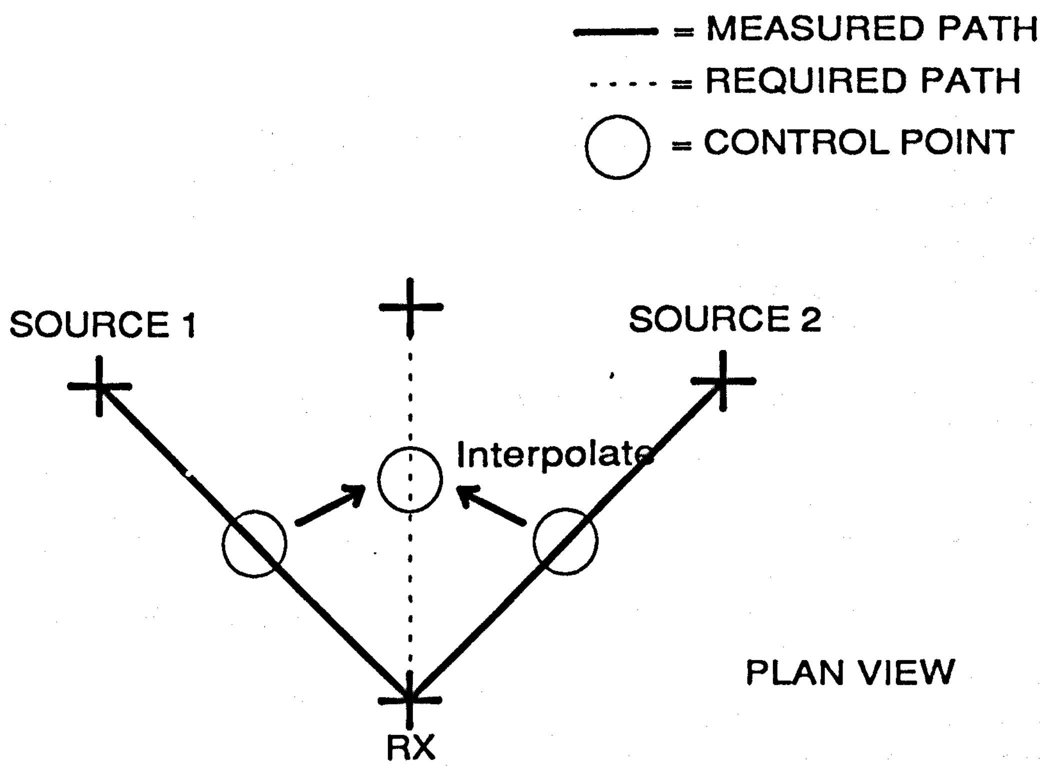
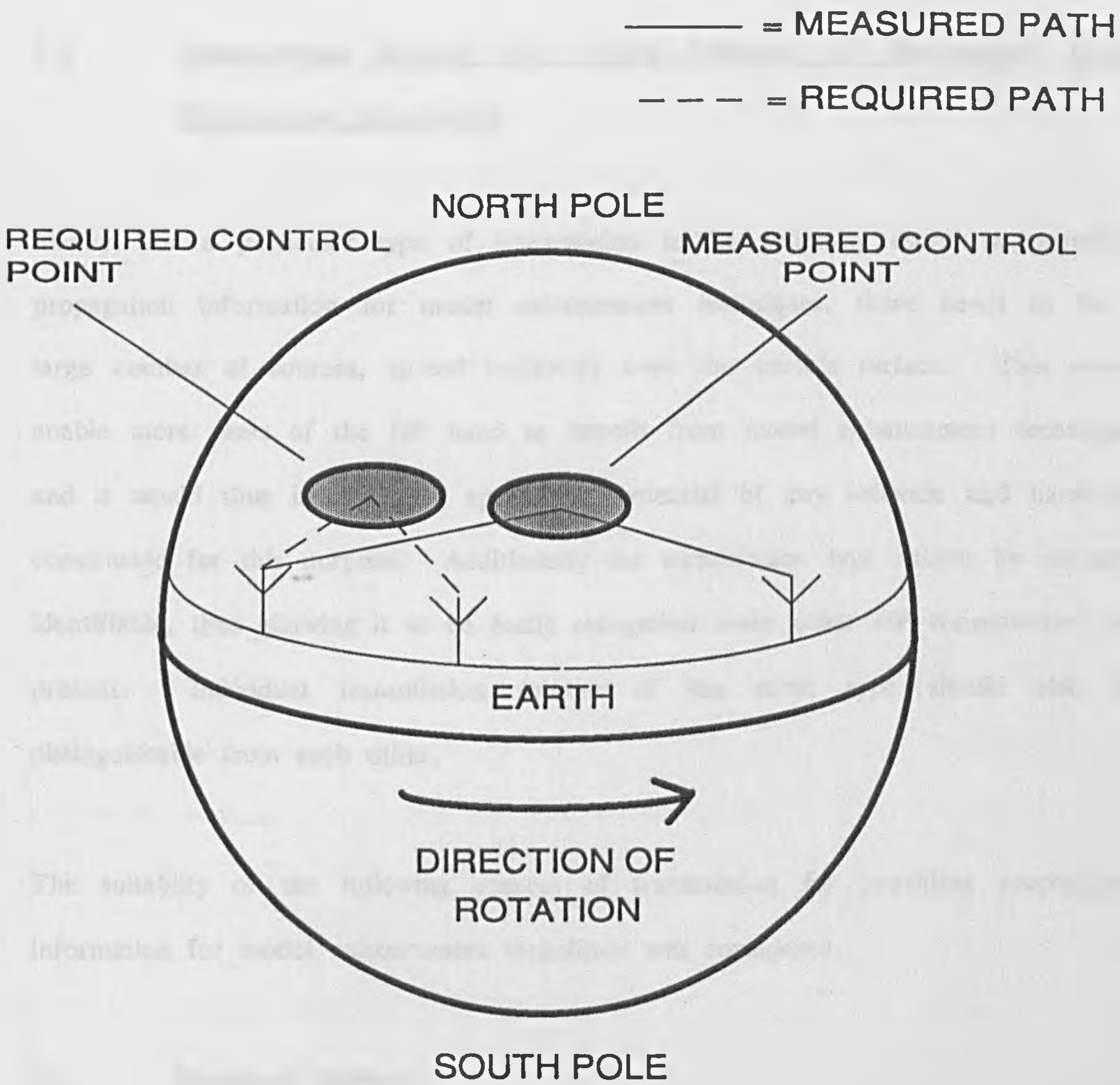


FIGURE 7.2. IONOSPHERIC "EARLY WARNING"



an "early warning system" for HF users, and if it proved to be valid it could be used to provide accuracy enhancement information for a propagation model or in the limit, eliminate the need for a model altogether. Only a limited number of users would be able to benefit from this technique as it requires there to be signal sources to the East of the path of interest.

7.3 Transmissions Suitable For Passive Provision Of Propagation Model Enhancement Information

Ideally, for a particular type of transmission to be generally useful in providing propagation information for model enhancement techniques, there needs to be a large number of sources, spread uniformly over the earth's surface. This would enable more users of the HF band to benefit from model enhancement techniques and it would thus increase the application potential of any software and hardware constructed for this purpose. Additionally the transmission type should be uniquely identifiable, thus allowing it to be easily recognised when other HF transmissions are present. Individual transmission sources of the same type should also be distinguishable from each other.

The suitablity of the following sources of transmission for providing propagation information for model enhancement techniques was considered:

(i) Broadcast Stations:

There are a large number of these, spread widely around the world and across the HF spectrum. Hence they should be able to yield information which would be of use to HF communication systems operating over most areas of the globe. However, it is difficult to distinguish between many of the different broadcast transmissions as they have no obviously unique features.

(ii) **Beacons:**

Beacons by their nature are easy to identify. However they can only provide ionospheric information pertinent to a few paths due to the fact that they are limited in number.

(iii) **Chirpsounders:**

There are over 40 of these transmitters operating from sites all around the world. They have a frequency versus time characteristic which makes them easy to detect. Also, they operate on a precise start-time schedule which enables each particular source to be identified uniquely.

(iv) **Time Standards:**

These are few in number but they are relatively easy to identify and detect.

It was decided that chirpsounder transmitters provided the best combination of geographical and numerical availability, with ease of detection and transmission source identification. Thus, for the purpose of the work performed on propagation model accuracy enhancement techniques, it was decided to base the propagation parameter measurements on received chirpsounder signals.

7.4 Chirpsounders

Chirpsounders are normally employed in an oblique incidence sounding mode. They transmit a low-level (100W ERP) tone which is scanned linearly upwards in frequency across the whole, or part, of the HF band, with respect to time. Details of the chirpsounder ionospheric measurement system are presented in (Barry, 1966). Chirpsounders facilitate the measurement of ionospheric path parameters, such as MUF, and the derivation of the mode structure that exists on a path, at every five

FIGURE 7.3. BLOCK DIAGRAM OF THE BARRY CHIRPSOUNDING SYSTEM

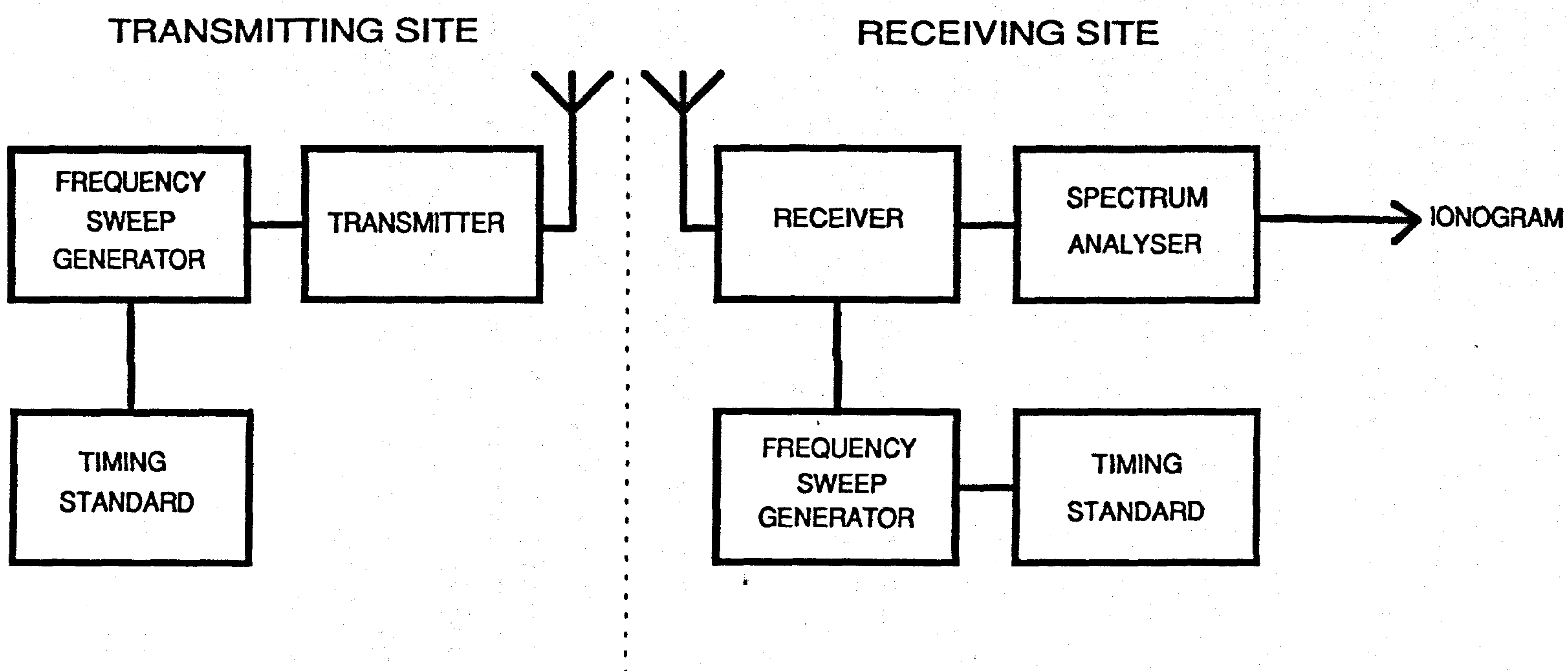
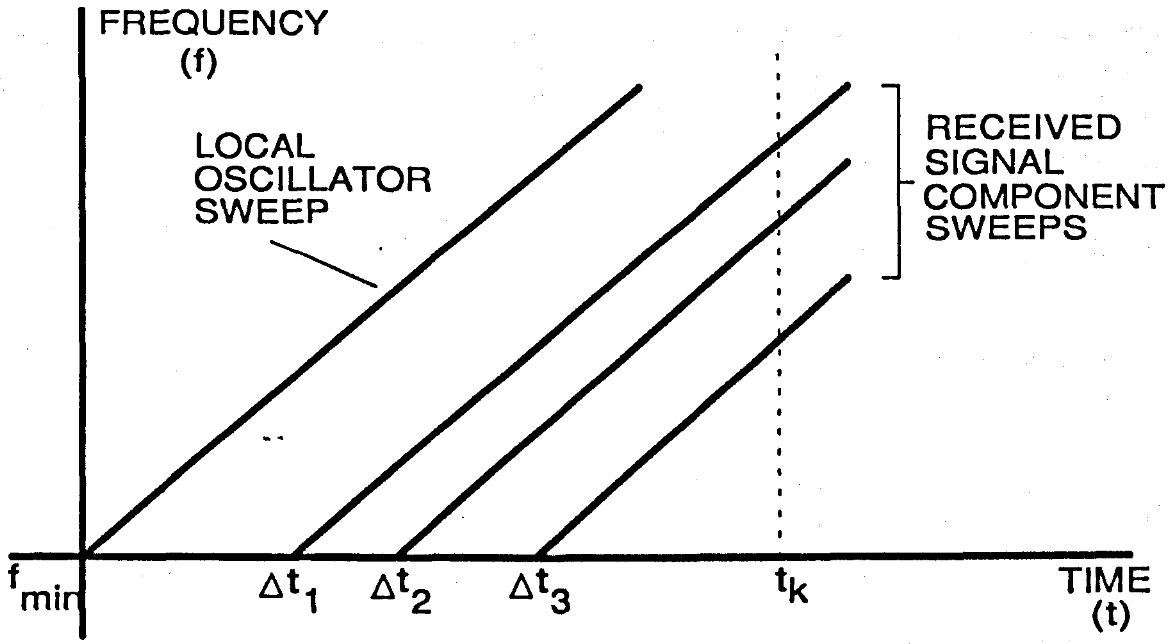


FIGURE 7.4. PRINCIPLE OF CHIRPSOUNDING



minute interval. The output from a chirp receiver is presented in the form of an ionogram.

A block diagram of the chirpsounder ionospheric measurement system described in (Barry, 1966) is shown by Figure 7.3. The chirp frequency sweep is linear with respect to time. The principle of the sounding of ionospheric radio channels using chirpsounders is described below with reference to Figure 7.4 (Darnell, 1983).

In multi-path propagation conditions several weighted versions of the transmitted signal will be received, as shown by Figure 7.4. A correctly timed local oscillator (LO), which is itself a replica of the original transmitted chirp, is then mixed with the incoming waveforms and the difference frequency components are filtered off. These components are then subjected to spectral analysis, as is shown by Figure 7.3.

Considering Figure 7.4:

At time t_k, the frequency of the LO sweep is

$$f_{min} + t_k.df/dt \qquad (7.1)$$

whilst the corresponding frequencies of the received multi-path component sweeps are:

$$f_{min} + (t_k - \Delta t_1).df/dt \qquad (7.2)$$

$$f_{min} + (t_k - \Delta t_2).df/dt \qquad (7.3)$$

$$f_{min} + (t_k - \Delta t_3).df/dt \qquad (7.4)$$

After mixing with the LO, the frequency difference components are:

$$\Delta t_1.df/dt \qquad \Delta t_2.df/dt \qquad \Delta t_3.df/dt \qquad (7.5)$$

Thus the propagation delays of the various multi-path components are translated into frequency offsets. If the mixing process is linear then the relative amplitudes of the components will be preserved. Time dispersion, due to the distributed nature of the ionospheric refraction process will cause the received sweep components to be broadened away from the spectral lines. Thus, displaying the results of this process on a spectrum analyser will give a propagation mode profile which is equivalent to the channel impulse response. An ionogram can then be produced by taking a projection of the spectrum analyser output as a function of LO frequency.

A "full-band" sweep, covers the frequency range 2 to 30 MHz in 280 seconds. A "half-band" sweep (2 to 16 MHz) has the same duration and thus has half the sweep rate. Most chirp transmitters use the full-band sweep and transmit every five minutes. The details of all known chirp transmitters are given by (Controller Defence Communications, 1986). There are 44 chirpsounders listed in this directory, although it is known that there are many more in existence.

7.5 Chirpsounder Detection and Demodulation Methods

The conventional method of extracting HF path propagation information from chirpsounder transmitters involves the use of a purpose-built receiver, manufactured by the Barry Research Corporation. Full details of the operation of this unit are to be found in (Barry, 1966).

This device produces an ionogram as its output and it costs approximately £50k (source: Marlborough Communications Ltd.). The aim of the work described in this chapter is to use chirp transmissions to provide propagation model accuracy enhancement information for the frequency management routines embedded within the terminal of an automatic HF comunications system. Since a complete HF terminal can cost as little as £10-20k, it will not normally be cost-effective to include a conventional chirp receiver in each terminal of a network of stations. Hence it was with this cost consideration in mind that other methods of extracting propagation information from chirpsounders were sought.

Dispersive networks have also been used in the detection and demodulation of chirp transmissions. A chirp-based modem is described in (Gott et al, 1989) which uses dispersive networks for demodulation of data transmissions. These make use of a knowledge of the frequency versus time characteristic of the chirp to be detected. The required network is an all-pass filter with a frequency versus time characteristic that is the inverse of that of the chirp, ie it delays the low frequency portion of the chirp the most - assuming a low-to-high frequency sweep. In this way, all of the energy of the chirp transmission appears at the output of the network simultaneously, and thus the arrival of a chirp is signified by a large pulse appearing at output of the filter. The required magnitude and group delay response of the detection filter is as shown in Figure 7.5 for a nominal 3 kHz wide frequency sweep.

In order to produce information concerning the propagation conditions on a path, the filter would be connected to the audio output of a standard HF communications receiver. The receiver would then be tuned ahead of the chirp transmission, but in synchronism with the transmitter. The output of the network provides propagation information at each step when the chirp passes through the channel to which the receiver is tuned. By linking the data from all such steps across the band, a "rough" ionogram can be produced, in the manner shown by Figure 7.6.

FIGURE 7.5. DISPERSIVE NETWORK RESPONSES FOR CHIRP DEMODULATION

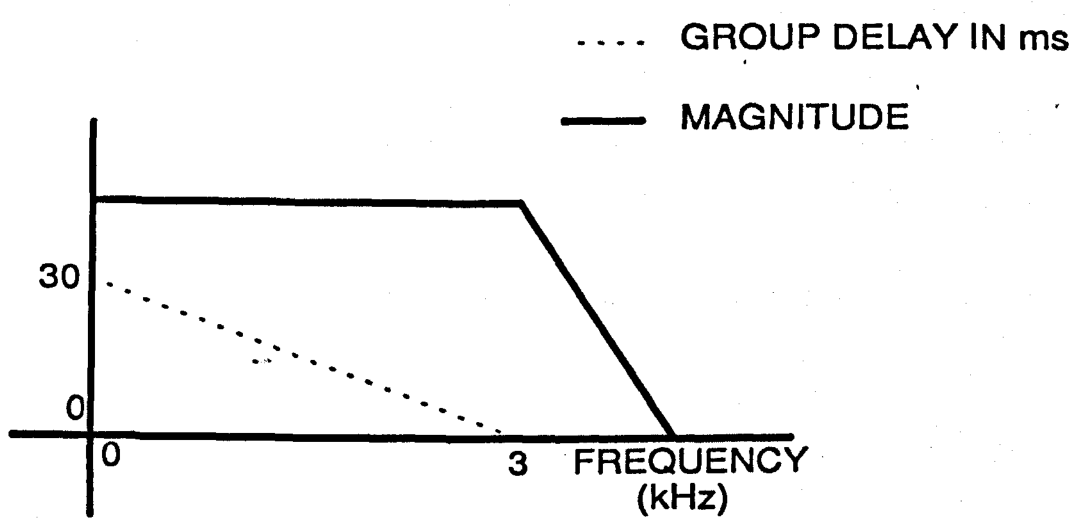
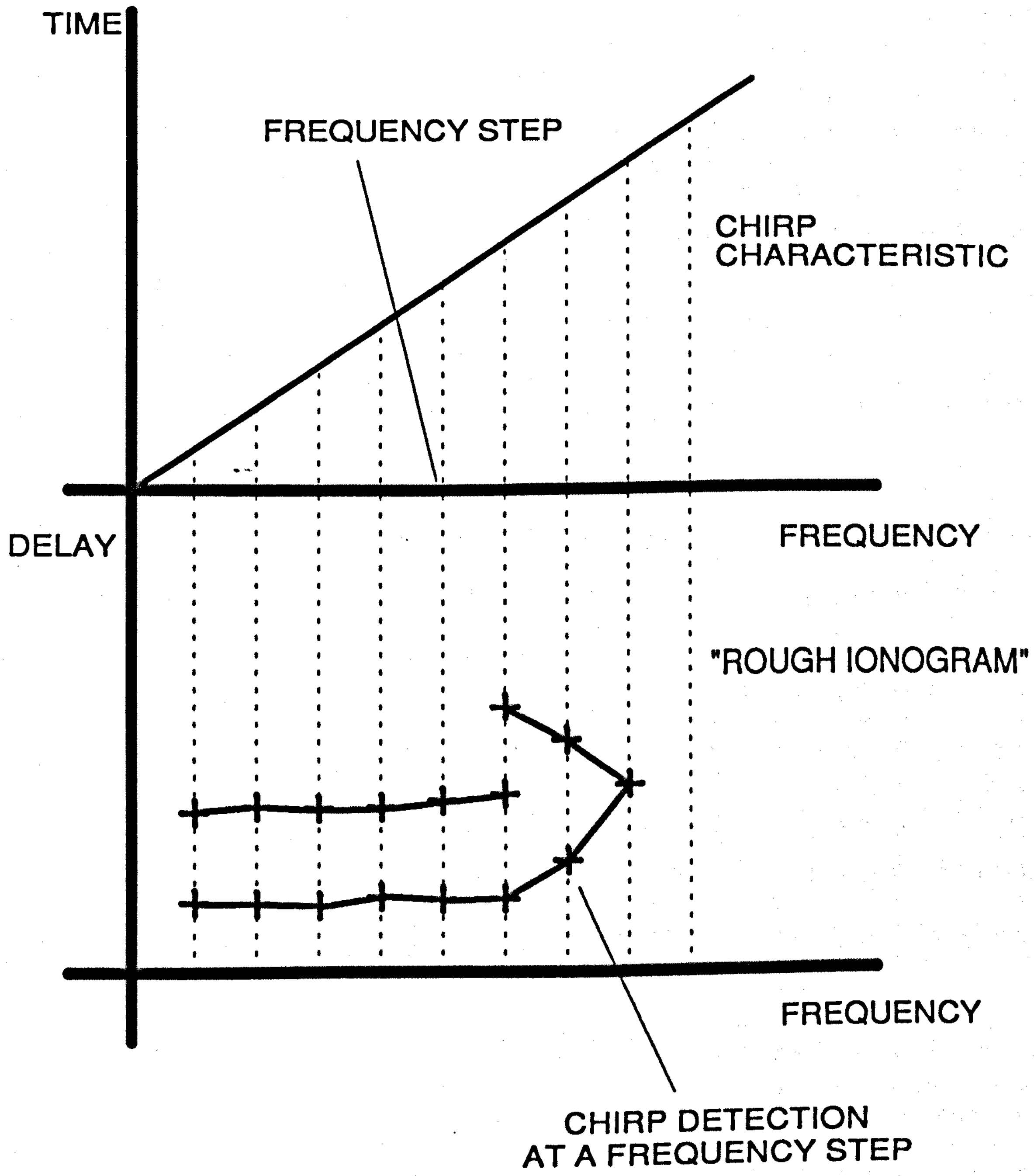


FIGURE 7.6. PRODUCTION OF A "ROUGH " IONOGRAM FROM STEPPED CHIRP DEMODULATION



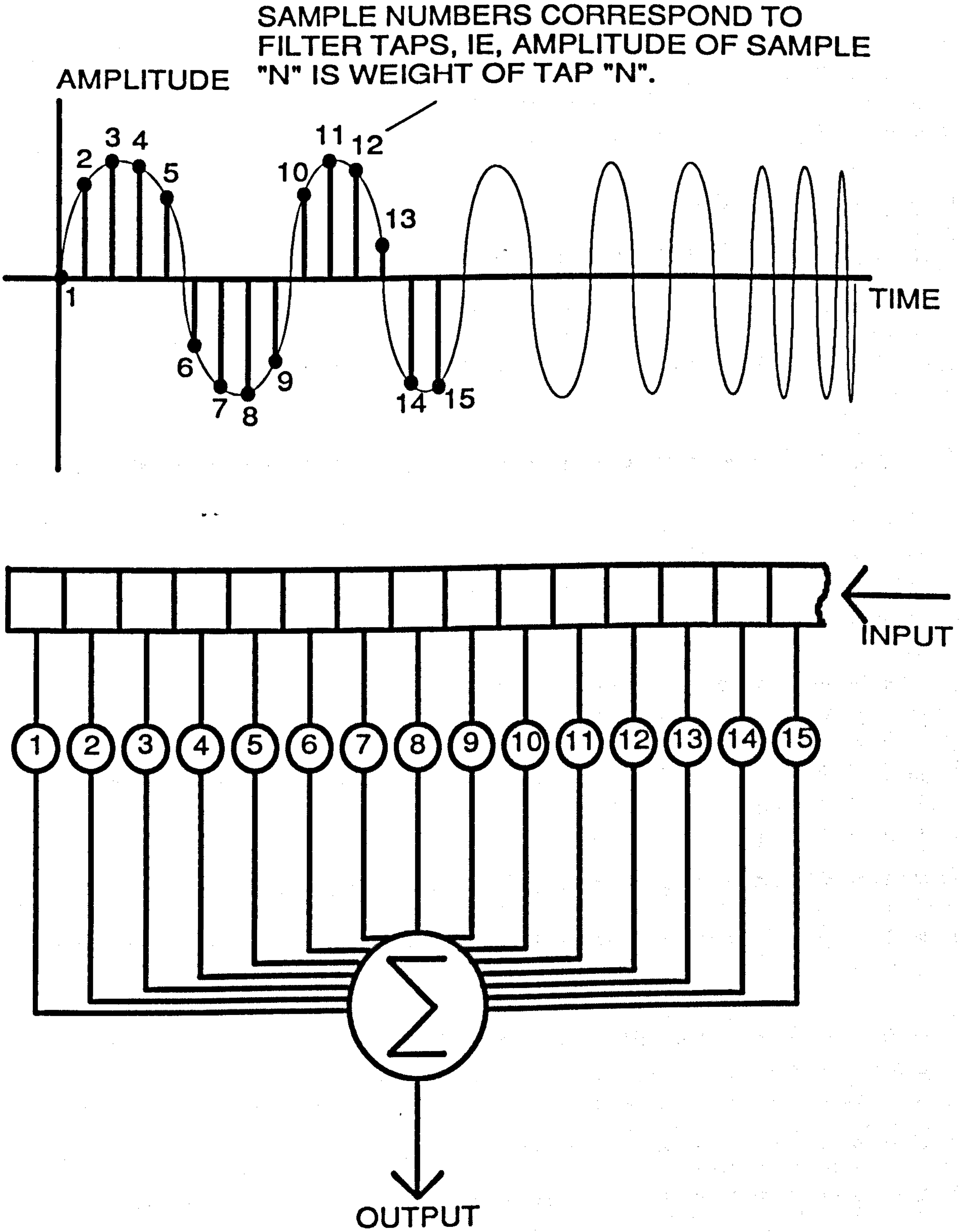
Chirp detection can also be effected via the use of a matched filter. This would take the place of the dispersive network in the above "step ahead" chirp demodulation scheme and it is potentially much simpler to implement. The filter taps have values equal to time domain samples of the chirp waveform as it passes through the receiver bandwidth. This method has the advantage that matched filters can be simply implemented on modern digital signal processors, such as the Texas Instruments' TMS 320 family. As such devices will be present in future, automatic HF communication systems, perhaps performing other signal processing functions such as filtering or spectrum analysis, no additional hardware cost would be incurred by the addition to the system of a chirp receiver constructed in this fashion. As a result of the ease of implementation and relative low-cost of this approach, it was decided to build a chirp detector using a matched filter. This would then be used in the "step-ahead" manner to enable ionospheric information to be extracted from chirpsounder transmissions.

7.6 Design And Implementation Of A Chirp Matched Filter And Chirp Detector

It was decided to implement the chirp matched filter in software on a TMS 320C25 development board as one such board had already been purchased for other aspects of the research, and the architecture of the TMS 320 was designed with digital signal processing operations in mind (Texas Instruments, 1986). It was felt that this software-based approach would be the most flexible and efficient.

The board used ran whilst resident in an expansion slot on an IBM pc XT clone, allowing programs to be developed and assembled on the pc, and then to be downloaded to the board for testing.

FIGURE 7.7. STRUCTURE OF THE CHIRP MATCHED FILTER



In order to avoid spending time learning TMS 320 assembly language, it was initially thought that the filter would be coded in a high-level language, such as C. However, discussions with staff at Plessey (Roke Manor) who had used the board, and with its manufacturers, revealed that existing language compilers for the TMS 320 produce code which runs, on average, six to eight times slower than hand-written assembly code, for the same application. As the time required by the processor to perform the necessary filtering was at this stage unknown, it was decided to implement the filter using assembly code. Considerable time was thus spent learning TMS 320C25 assembly language.

The initial structure of the chirp matched filter was as shown in Figure 7.7. The incoming chirp waveform is sampled (at the same sample rate as that used to produce the tap weights) and the samples are passed along the delay line. When a match occurs, ie a chirp is present, a large pulse appears at the output of the filter. Hence by observing this output as a function of time it is possible to specify when a chirp has occurred. Additionally, the occurrence of multi-path conditions would be signalled by several pulses of differing heights occuring within a short time interval.

For an automatic chirp logging system the filter output could then be used as a source of interrupts for another microprocessor, such as that present in the pc itself. In this way the processor logs a chirp occurrence whenever it receives the specified interrupt (multi-path information may be lost, however, due to the reaction time of the TMS to pc interrupt system).

A 3 kHz bandwidth was initially assumed for the design of the chirp filter since this corresponds to the reception bandwidth of most HF communications receivers. Therefore a sample rate of greater than 6 kHz was necessary in order to avoid problems with aliasing. The duration of a chirp in this bandwidth (which also governs the length of the delay line of the filter) is 30 ms. A sample rate of 8 kHz was chosen which meant that a 240 tap filter was necessary.

The tap weights are generated by the pc and downloaded to the TMS 320's memory. With an 8 kHz sample rate, the TMS 320C25 is able to perform 1250 machine cycles between samples. The filter program requires approximately 240 machine cycles per sample input. The choice of assembly language was thus justified, bearing in mind the execution time overheads incurred by using a high-level language.

An algorithm to generate the filter tap weights was derived, with the phase of the chirp waveform being preserved between frequency increments and hence filter taps. The derivation of the filter tap weight generation algorithm is as follows:

The instantaneous phase of a sinewave is given by :

$$\text{phase} = \int_0^T \text{frequency}\ dt \qquad (7.6)$$

where T is the point in time under consideration.

Amplitude of a sinewave is defined as :

$$\text{Amplitude} = \text{sine}\ (\ \text{phase}\ ) \qquad (7.7)$$

The frequency of a chirp waveform increases linearly with time. Therefore :

$$\text{frequency} = k.t \qquad (7.8)$$

where k is a constant.

Substitute equation 7.8 into equation 7.6, i.e. for a swept frequency sinewave :

$$\text{phase} = 2.\pi\int_0^T k.t.dt = \pi.k.T^2 \qquad (7.9)$$

Also, from equation 7.7:

$$\text{amplitude} = \sin (\pi.k.T^2) \qquad (7.10)$$

If the chirp waveform is sampled (frequency increases in steps of size f_{inc} at each time increment t_{inc}), and a scaling factor M is introduced, then equation 7.10 becomes :

$$A_i = M.\sin (\pi.f_{inc}.t_{inc}.i^2) \qquad (7.11)$$

where A_i is the instantaneous amplitude of the chirp waveform and i is the sample number ($0 \leq i \leq T/t_{inc}$). The values of the chirp matched filter taps and the samples of chirp waveform used by the chirp generator software were thus calculated using equation 7.11.

7.7 Testing Of The Chirp Matched Filter

In order to test the chirp matched filter, a cassette tape recording of a repetitive 30 ms portion of the chirp waveform was produced. The test was carried out using the pc and the TMS 320 board in the following manner.

The C program used to generate the chirp matched filter taps was altered to produce 1200 samples (instead of the original 240) of the same chirp waveform. These samples were then downloaded to the external data memory of the TMS 320 and a program was written in 320 assembly language which continuously wrote the samples to the digital-to-analogue (DAC) converter on the development board. The sample output rate used was 40 kHz. The aim of using a higher rate than that employed by the matched filter was to make the generated chirp appear almost analogue in nature to the filter, and hence the tests performed using the recording would be more realistic. Accurate synthesis of a true analogue chirp waveform with

FIGURE 7.8.

FIGURE 7.8a. OUTPUT OF CHIRP GENERATOR

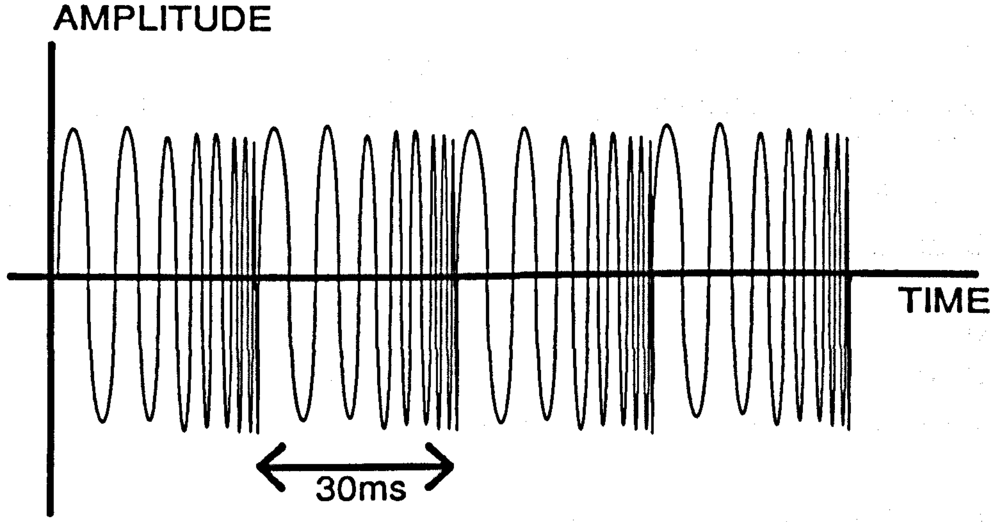


FIGURE 7.8b. CHIRP RECORDING APPARATUS

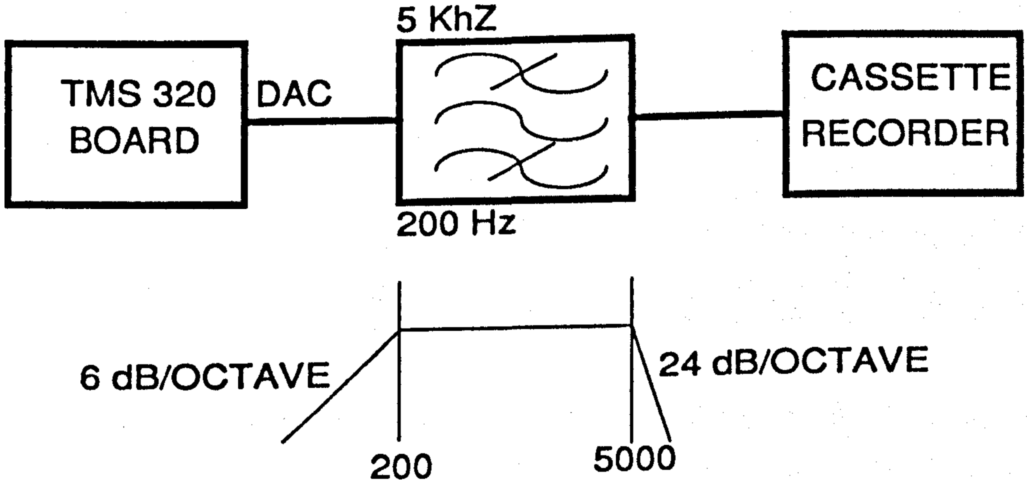


FIGURE 7.9.

FIGURE 7.9a. CHIRP DETECTOR TEST APPARATUS

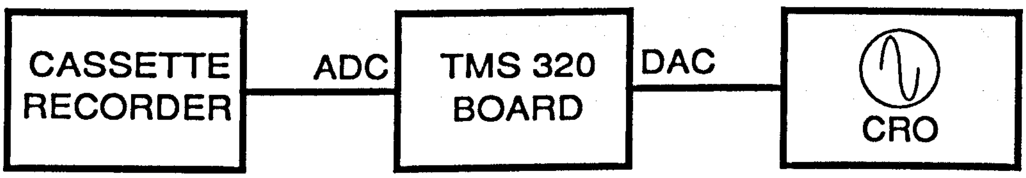
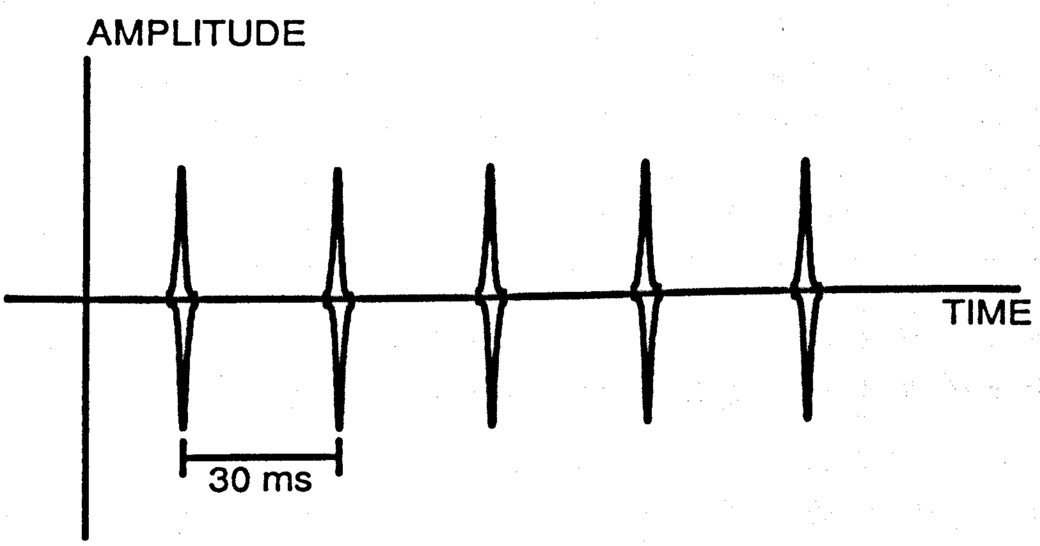


FIGURE 7.9b. RESPONSE OF CHIRP MATCHED FILTER



the same parameters as the chirps produced by the sounder transmitters would be more difficult to achieve in practice. The sample output rate and the number of samples were calculated so that a 30 ms portion of chirp waveform would be produced. This represents the length of chirp that would occur in a 3 kHz receiver bandwidth.

The resulting output was as shown in Figure 7.8a. The chirp generator was connected to a cassette tape recorder through a band-pass filter, as in Figure 7.8b and a recording of chirps was produced for testing purposes.

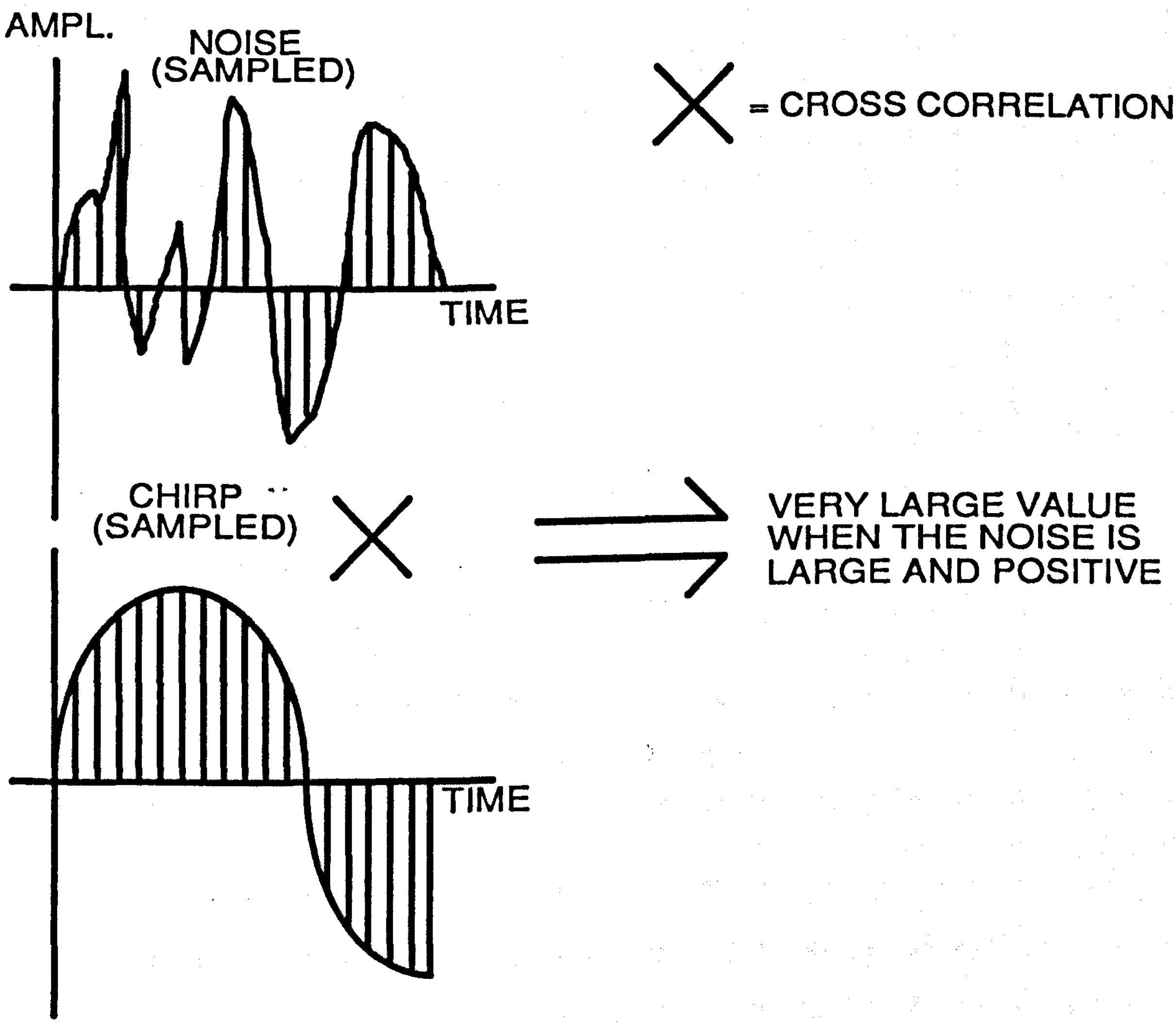
The chirp matched filter software was altered so that its output was compared with two threshold levels, set at half the maximum positive and negative excursions of the filter output. If the filter output exceeded one of these thresholds then this output was passed on to the DAC. In this way a change in level of the output of the threshold stage indicated that a chirp had been detected.

To test the combined filter and threshold software, termed the "chirp detector", the apparatus of Figure 7.9a was set up. The tape recorder was switched on and the response of the chirp matched filter alone was as shown by Figure 7.9b. The threshold software also appeared to be functioning correctly with a change in output being produced for each time a chirp occurred. Multiple output level changes were observed as the match became close.

Subsequently, the chirp detector was tested with on-air chirps. This was achieved by connecting the audio output of an HF receiver to the input of the matched filter and waiting for a chirpsounder to pass through the receiver bandwidth.

The threshold levels at the output of the matched filter were re-adjusted to be just greater than the peak value of the filter output with no chirps present, ie just above (for the positive threshold) and below (for the negative threshold) the

FIGURE 7.10. CHIRP MATCHED FILTER NOISE RESPONSE



response of the filter due to the background noise from the receiver.

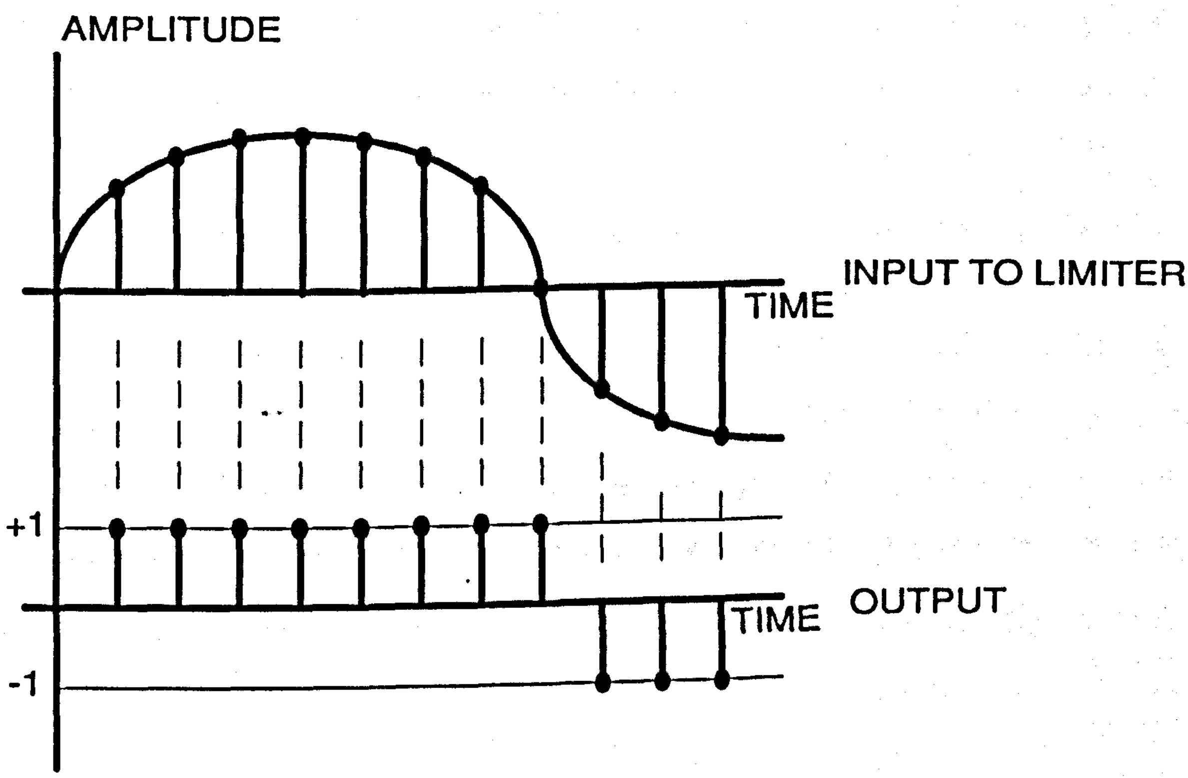
The Chirpsounder Directory (Controller Defence Communications Network, 1986) was used to select a chirpsounder for trials, the aim being to find one which was geographically close so that the received signal strength would be relatively high. RAF Milltown in Scotland was used as the chirp source for trials. The receiver was tuned to a frequency which was less than the predicted MUF for the path from the sounder site to Hull (this figure was derived from a prediction model based upon (CCIR, 1978)) and which was relatively quiet. When the time of intercept approached, the chirp detection software was run and the output of the detector and receiver audio output was monitored.

The incidence of a chirp was signalled by a short duration, swept frequency tone being heard on the audio output of the receiver. The detector reliably changed output level, ie signalled the arrival of a chirp, when there was one. Also it reliably changed level when a large noise spike occurred (such as that generated by a coffee-maker switching on and off!). Further modification to the detector was thus deemed to be necessary.

A large noise spike would have a peak amplitude much larger than the background noise coming from the HF receiver. It is feasible, therefore, for the product of such a spike and the larger amplitude filter taps to produce a positive response from the chirp detector. Figure 7.10 illustrates how this can happen.

In order to eliminate this problem, it was decided to encode only the phase information of the chirp waveform in the detector; this was achieved by limiting the chirp matched filter taps and the filter input samples in the manner shown by Figure 7.11.

FIGURE 7.11. METHOD OF LIMITING FILTER TAPS AND INPUT



The same on-air tests were tried once more and it was found that the noise immunity of the detector was much improved. It only responded to chirps and not to mains spikes and glitches. Chirps from a number of locations were heard and reliably detected, including those from Chelveston, Inskip, Milltown, Blandford, Poole, Locking, Cobbett Hill, Norway, Gibraltar, Hong Kong and Italy. It should be noted that the antenna used in these initial trials was an arbitrarily oriented 10 m length of wire. Many more chirpsounders were heard and detected once a broadband horizontal HF antenna was installed at the receiving site.

7.8 A Chirp-Based Propagation Estimator

7.8.1 Introduction

The overall aim of the work described in this chapter was to provide update and enhancement information for propagation analysis and prediction algorithms. In order to enhance the accuracy of a propagation model, in the manner described in Section 7.2, it is necessary to measure a parameter for an ionospheric path, so that a comparison between predicted and measured values, and the subsequent adjustment of the propagation prediction algorithm, can be made.

Accurate information regarding the MUF over a given path is a pre-requisite for effective HF frequency management. In an attempt to provide accurate MUF data for HF frequency management systems, the chirp detector described in the Sections 7.6 and 7.7 has been incorporated into a propagation estimation system, to be used in enhancing the accuracy of propagation models. The estimator uses the "step ahead" modus operandi detailed in Section 7.5. The word "estimation" is used, rather than "measurement", as the system that was produced had a restricted resolution.

FIGURE 7.12. BLOCK DIAGRAM OF THE PROPAGATION ESTIMATOR

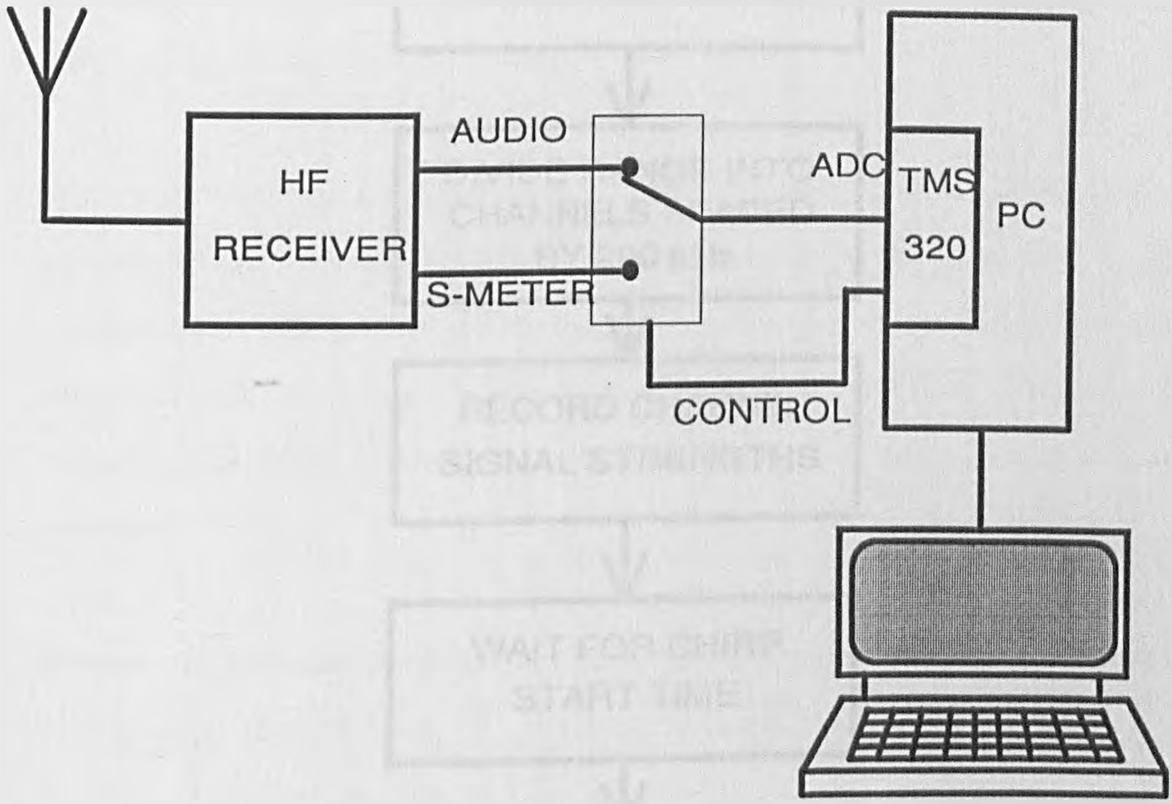
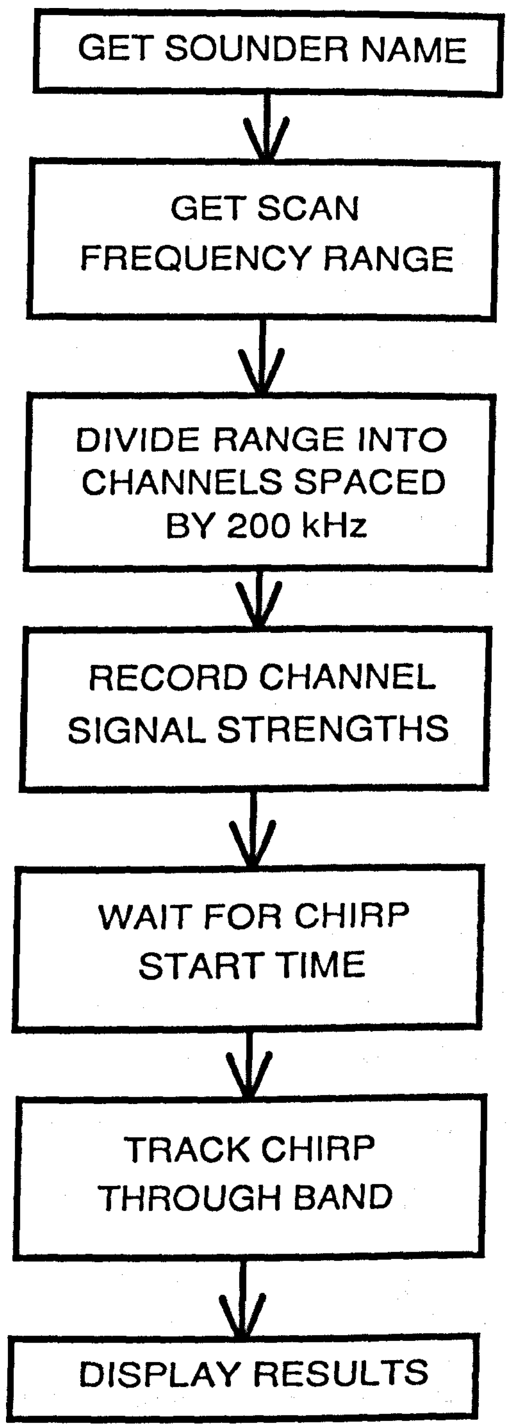


FIGURE 7.13. SEQUENCE OF OPERATIONS OF THE PROPAGATION ESTIMATOR



A block diagram of the estimator is shown in Figure 7.12. The hardware consists of a broadband HF antenna, an amateur grade HF receiver, an IBM pc XT clone, and a TMS 320C25 development board.

The sequence of operations of the system is as shown in Figure 7.13. The user enters the name of the chirpsounder to be scanned. The frequency range over which chirp tracking is to take place is then keyed in and the pc then divides this range into channel frequencies spaced apart by 200 kHz. This spacing value was chosen as it would allow a two second window for chirp detection at each step: this was thought to be sufficient to cater for any likely timing differences between the chirp transmitter and the monitor receiver.

The system then logs the signal strength present in each of the channels, and it eliminates those which are too noisy for chirp reception to be successful. After synchronisation with the chirp transmitter is achieved, the system follows the sounder through the entered frequency range, logging the chirp occurrences as it proceeds. The results of the scan are displayed visually and may be output in hard copy form.

A more detailed description of the signal strength assessment and chirpsounder tracking stages of operation of the propagation estimator now follows.

7.8.2. Signal Strength Assessment

After the user-defined frequency range is divided into channels spaced apart by 200 kHz, the signal strength on each channel is assessed. The aim of doing this is to eliminate noisy channels from the frequency list to be used for chirpsounder tracking and to provide soft-decision information for the user, when trying to estimate the MOF at the end of the system's operation.

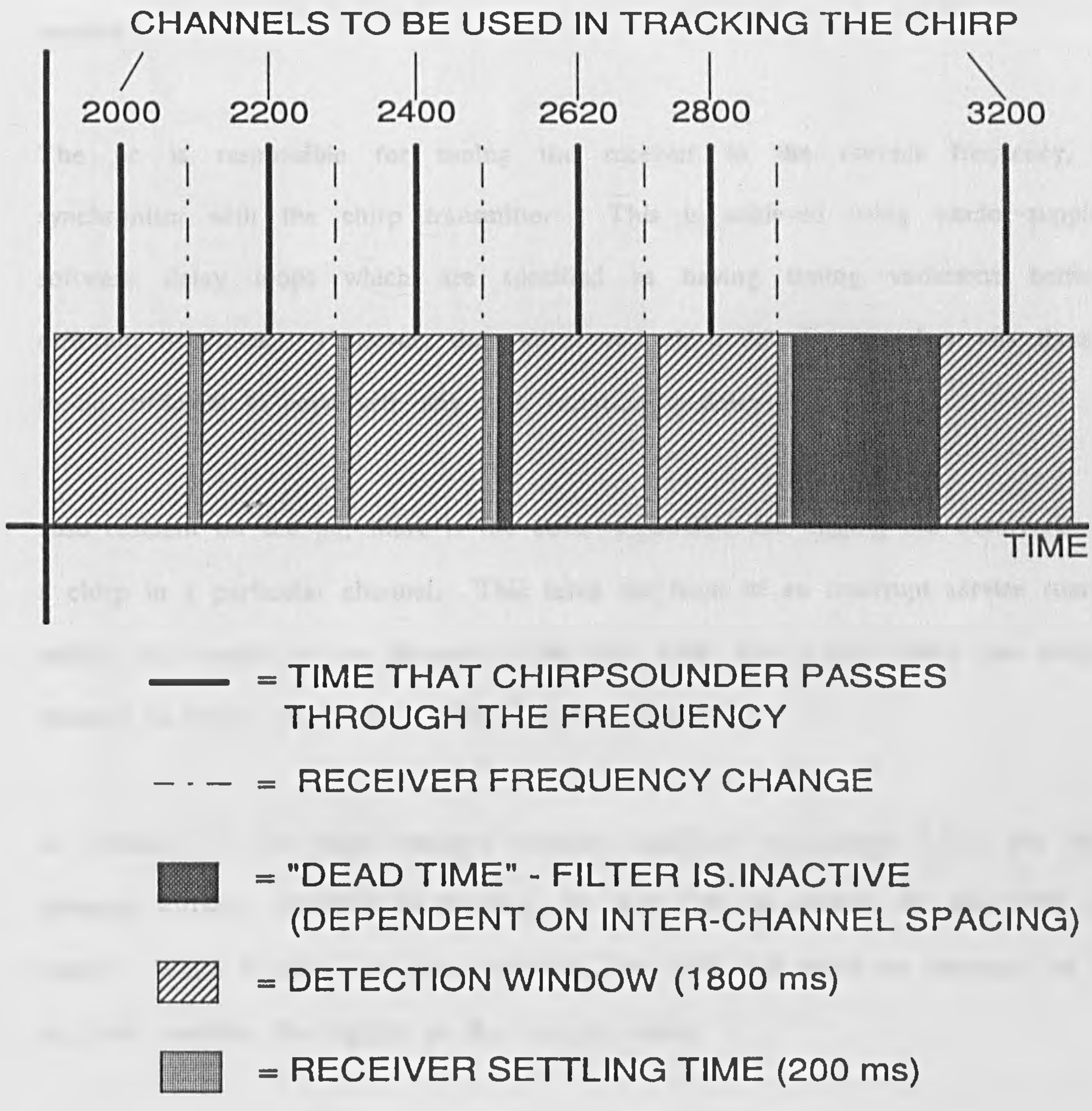
The signal strength measurement is performed by the TMS 320 board. It takes 256 samples, over a period of one second, of the voltage being fed to the S-meter of the receiver. This average value is passed back to the pc as being the signal strength in the current channel. The pc then decides whether or not the channel is sufficiently quiet, the threshold for this decision being set by trial and error. If a channel is found to be too noisy then two further attempts will be made, at frequencies 10 and 20 kHz higher than the original, to find one which is suitably quiet, before that frequency step is deemed to be too noisy for matched filter detection of the chirp to occur. If this proves to be the case, it is then removed from the list of channels to be scanned for in-channel chirpsounder occurrences.

At the end of the signal strength assessment, the pc contains a list of frequencies which are thought to be suitable for successful chirp detection to take place. An absence of a chirp in a channel should therefore be due to propagation limitations only; thus the signal strength assessment stage enhances the accuracy of the overall results from the system.

7.8.3 Chirpsounder Tracking

After assessing the signal levels in the channels, the system waits until the chirpsounder is due to pass through the first frequency. The time for commencing chirp tracking is calculated from data stored on the pc's hard disk, which has been taken from the Chirpsounder Directory (Controller Defence Communications Network, 1986), and the frequency value of the first channel in the list to be scanned. The internal clock on the pc is compared with the projected start time until they are equal and then the tracking of the sounder over the specified frequency range (minus noisy channels) begins.

FIGURE 7.14. TIMING DIAGRAM FOR PROPAGATION ESTIMATION SYSTEM



A precise timing diagram of the chirp tracking phase of the system is given in Figure 7.14. As can be seen from this diagram, the system automatically adjusts when the frequency difference between channels is not equal to 200 kHz; such inequality would be caused by the signal strength assessment phase adjusting the list of frequencies to be scanned for chirps, due to high levels of interference. It is necessary for the filter to be inactive sometimes to avoid detection of other chirp sources (the closest spacing between individual chirpsounder start times is 2 seconds).

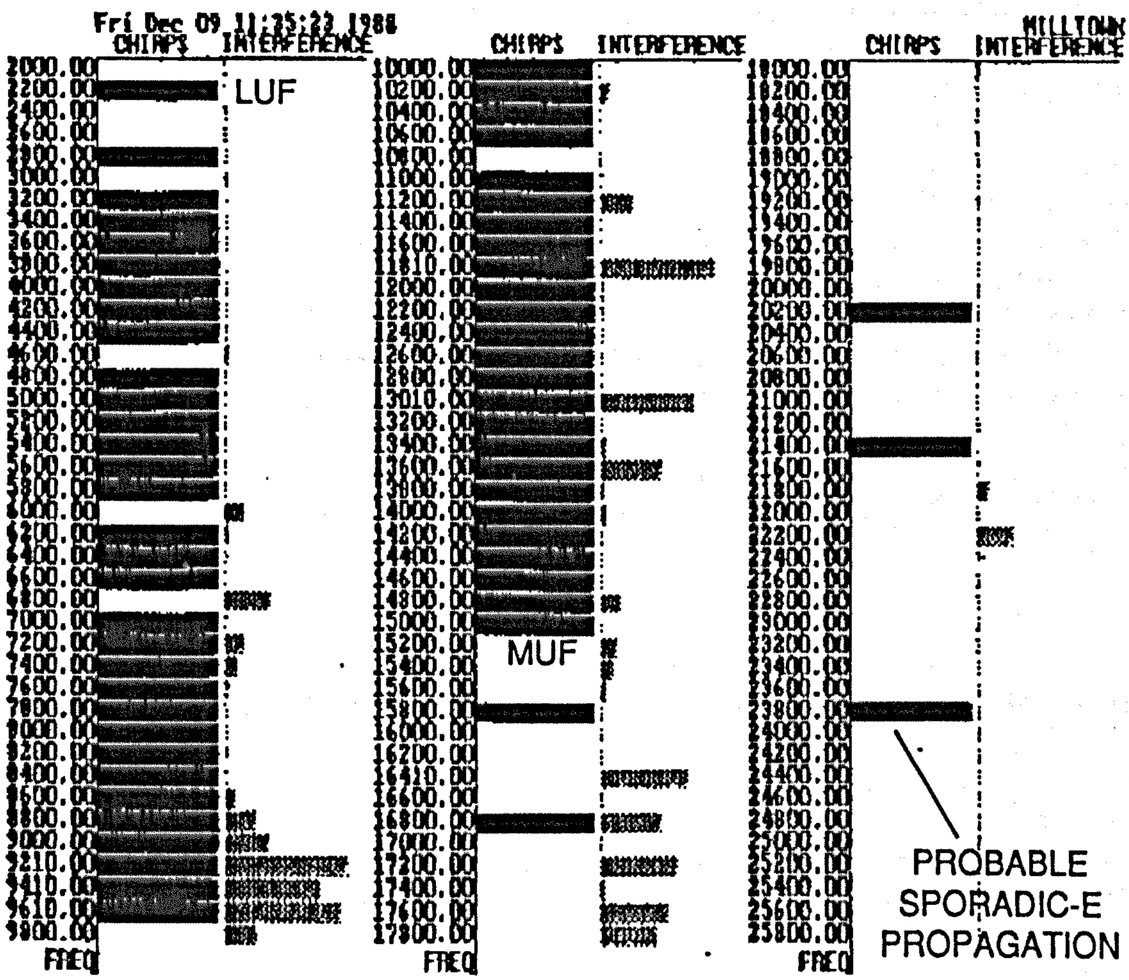
The pc is responsible for tuning the receiver to the correct frequency, in synchronism with the chirp transmitter. This is achieved using vendor-supplied software delay loops which are specified as having timing variations between different machines. However, the two-second chirp detection window was thought to be sufficient to cope with any such variation in delay loop length.

Also resident on the pc, there is the code responsible for logging the occurrence of a chirp in a particular channel. This takes the form of an interrupt service routine which, on receipt of an interrupt from the TMS 320 board, labels the current channel as being one where a chirp has been detected.

In addition to the signal strength routines described in Section 7.8.2, the chirp detector software (detailed in Sections 7.6 and 7.7) is resident on the TMS 320 board. When a chirp has been detected, the TMS 320 sends an interrupt to the pc, thus enabling the logging of the chirp to occur.

The results of both the signal strength assessment and the chirp tracking are displayed visually in a graphical format on the pc screen. An example of the output produced is shown in Figure 7.15.

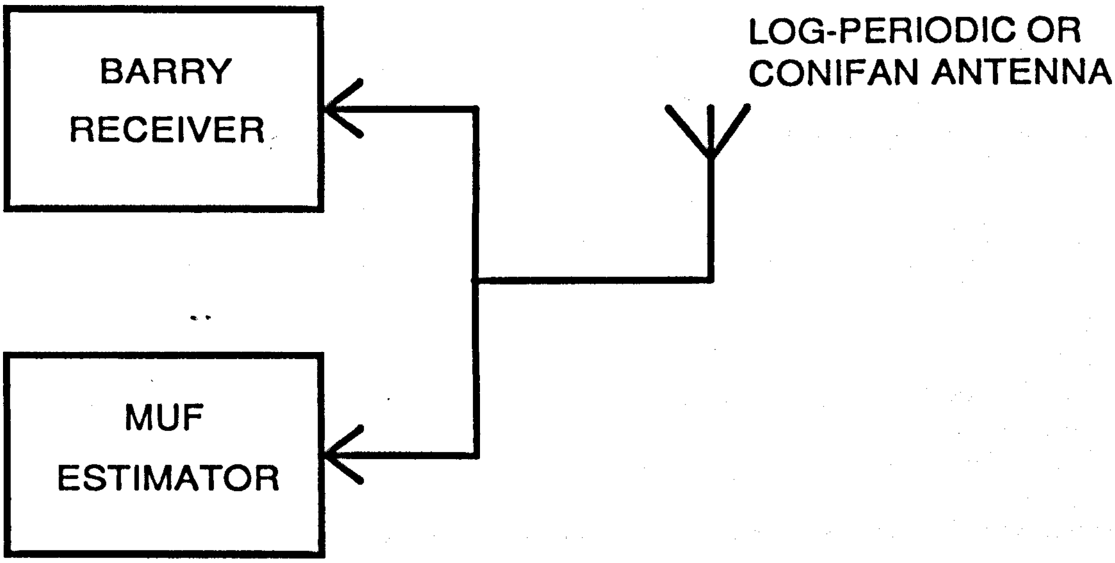
FIGURE 7.15. EXAMPLE OUTPUT FROM THE PROPAGATION ESTIMATOR



SOLID BARS REPRESENT A CHIRP OCCURRENCE IN A CHANNEL

SHADED BARS REPRESENT THE RELATIVE INTERFERENCE LEVEL EXPERIENCED DURING THE INTERFERENCE PRE-SCAN

FIGURE 7.16. CHIRP COMPARATIVE TRIALS APPARATUS



7.9 Comparative Trials Of The Propagation Estimator With A Barry Research Chirpsounder Receiver

The aims of these trials were:

i) Validation of the propagation estimator.

ii) Determination of the accuracy of the MOF estimates produced.

iii) Identification of any additional information which could be extracted from the output of the propagation estimator.

The trials were performed at the Royal Aerospace Establishment's Cobbett Hill receiving site in Hampshire. To enable the performance of the propagation estimator to be compared directly with that of the Barry receiver, both systems were connected to the same antenna, as shown in Figure 7.16.

Chirpsounders from four locations were tracked: RAF Milltown in Scotland, Bodo in Norway, Akrotiri in Cyprus, and Port Stanley in The Falklands. Figure 7.17 shows a sample of the output produced by both systems in response to the same sounder at the same time. It is seen from these results that the output of the propagation estimator can be used to derive maximum observable frequency (MOF) and lowest observable frequency figures (LOF) for a path and, in some cases, it registered sporadic-E mode propagation (although without the Barry ionogram this would be difficult to identify on the propagation estimator output).

Figure 7.18 shows a plot of the Barry Receiver MOF and that produced by the propagation estimator, over the whole duration of the trials. The differences between the two plots are small and usually these can be explained by the presence of an interferer or by the limited frequency resolution of the estimator.

FIGURE 7.17

RESULTS OF COMPARATIVE TRIALS BETWEEN THE BARRY CHIRPSOUNDER RECEIVER AND THE PROPAGATION ESTIMATOR

CHIRPSOUNDER TRANSMITTERS USED:

BODO, NORWAY.

AKROTIRI, CYPRUS.

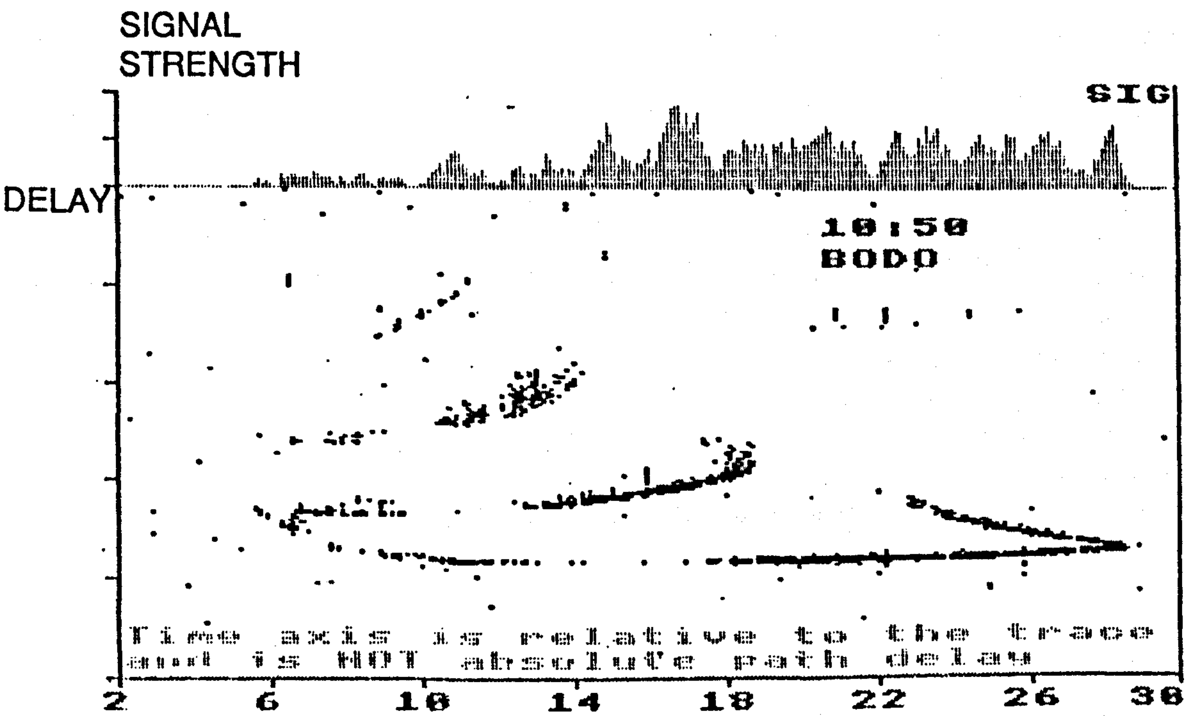
MILLTOWN, SCOTLAND.

PORT STANLEY, FALKLAND ISLANDS.

FIGURE 7.17a.

SOUNDER SOURCE: BODO, NORWAY

BARRY OUTPUT



PROPAGATION ESTIMATOR OUTPUT

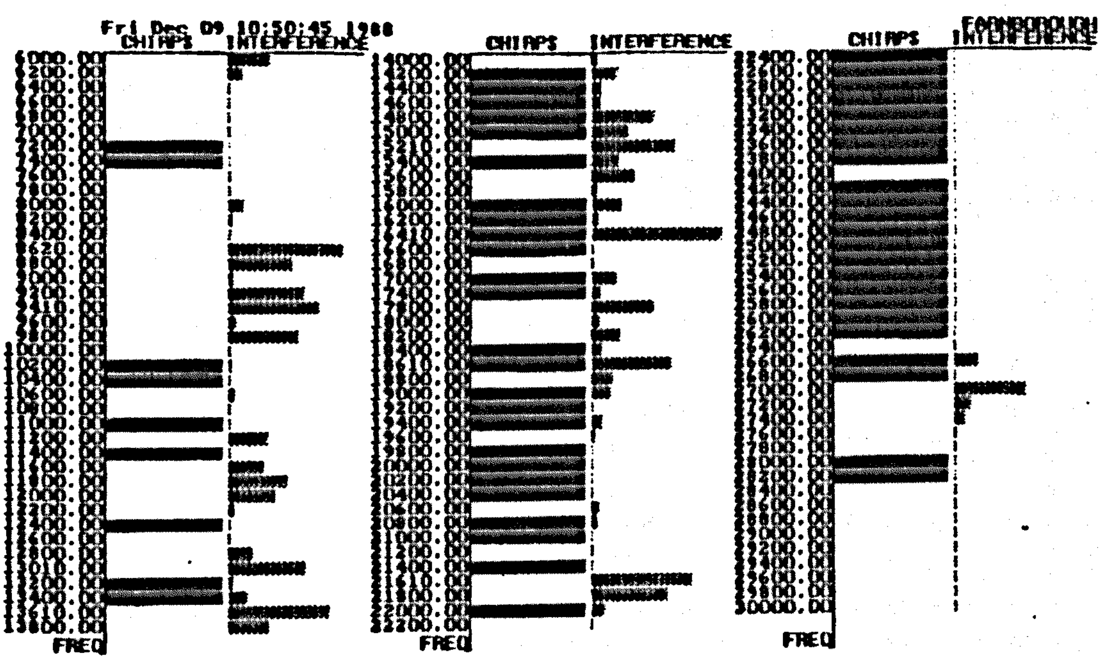
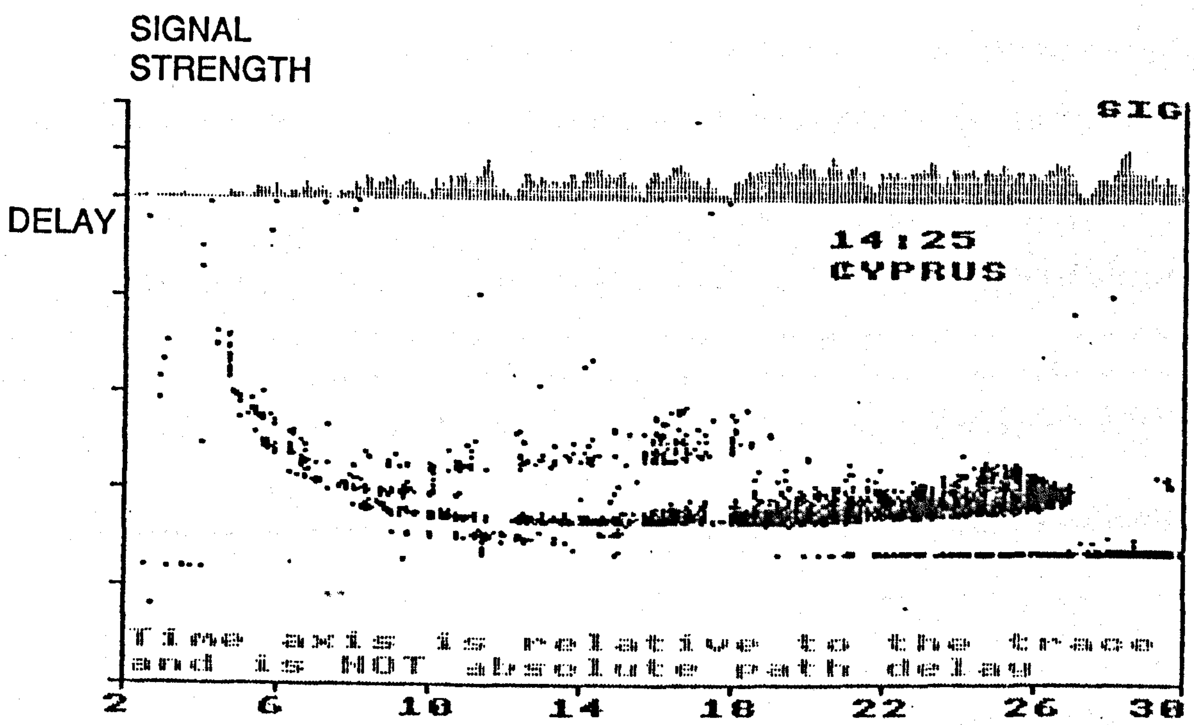


FIGURE 7.17b

SOUNDER SOURCE: AKROTIRI, CYPRUS

BARRY OUTPUT



PROPAGATION ESTIMATOR OUTPUT

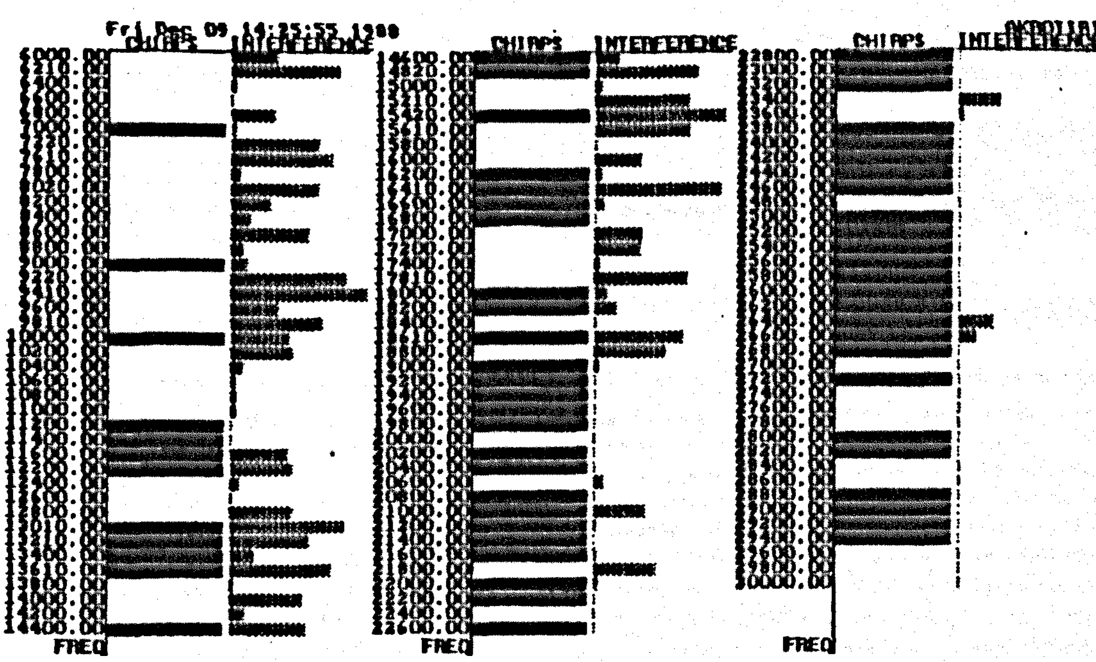
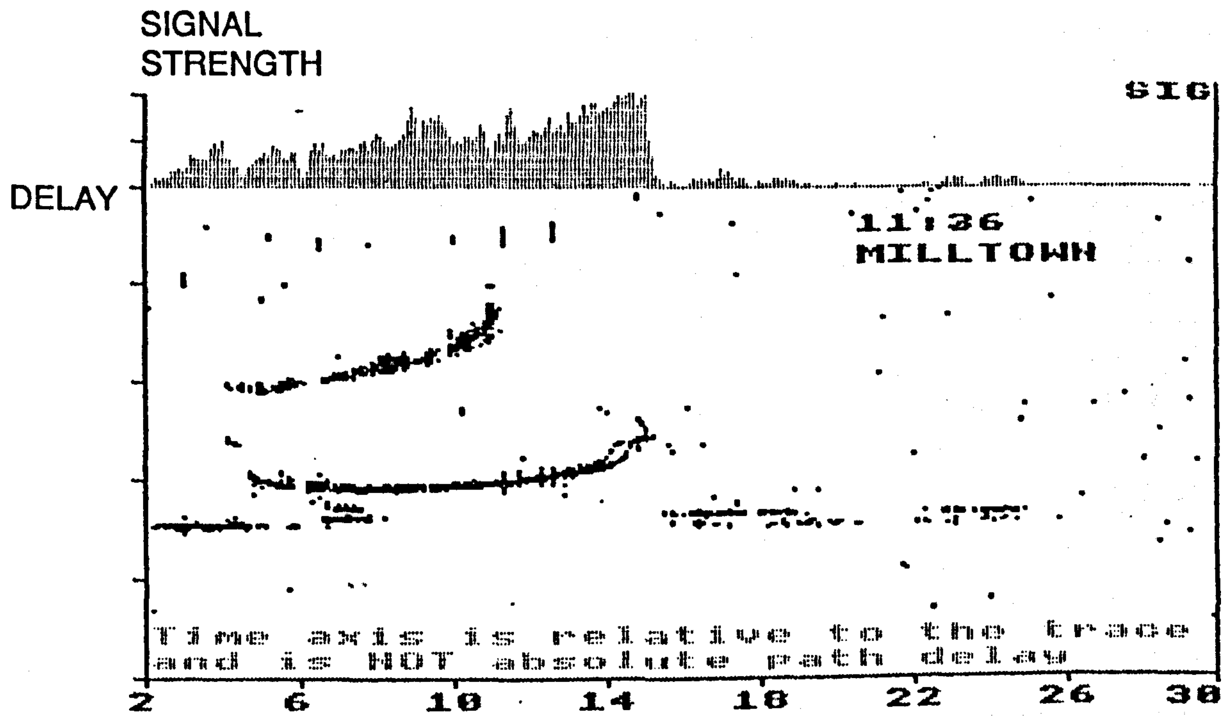


FIGURE 7.17c

SOUNDER SOURCE: MILLTOWN, SCOTLAND

BARRY OUTPUT



PROPAGATION ESTIMATOR OUTPUT

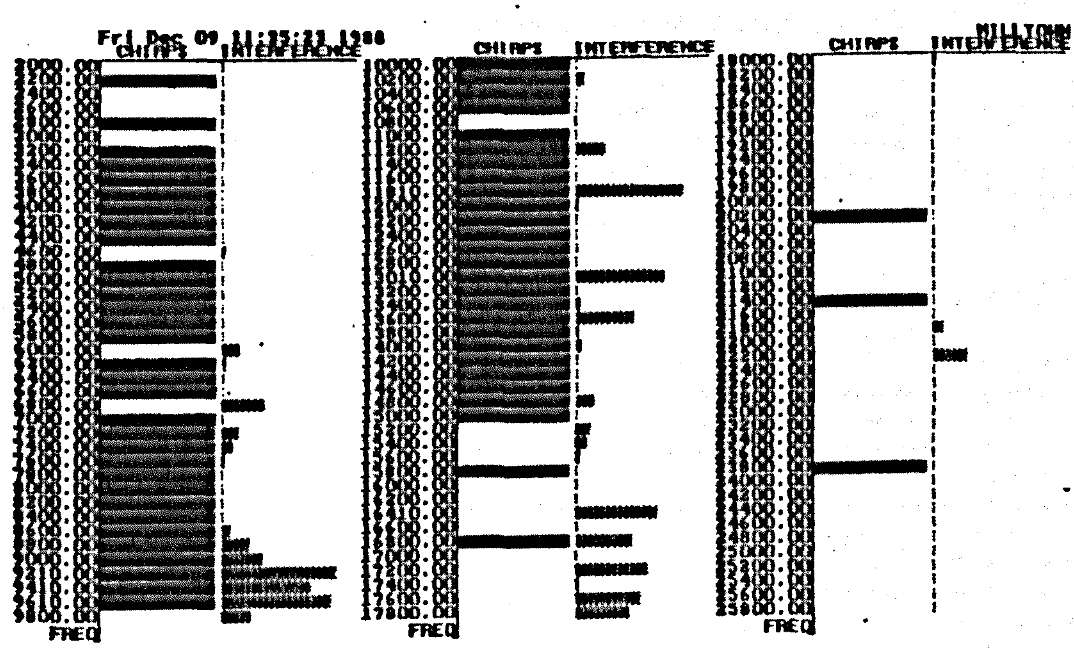
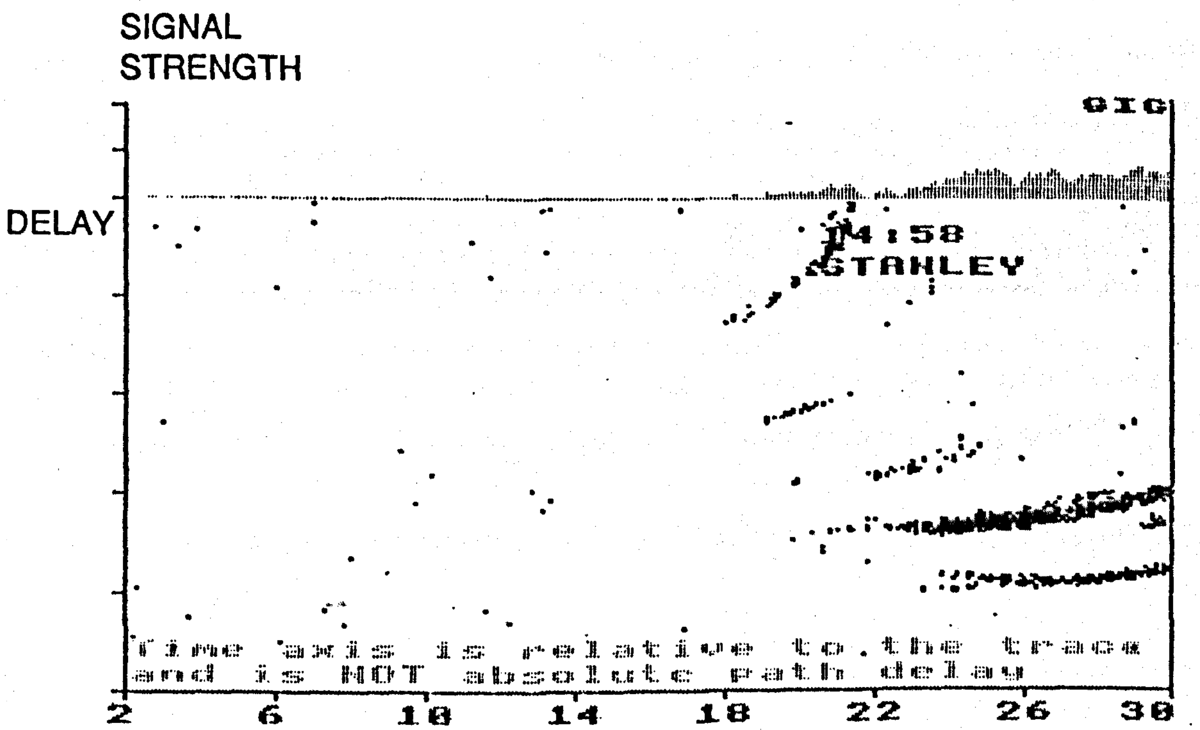


FIGURE 7.17d

SOUNDER SOURCE: PORT STANLEY, FALKLAND ISLANDS

BARRY OUTPUT



PROPAGATION ESTIMATOR OUTPUT

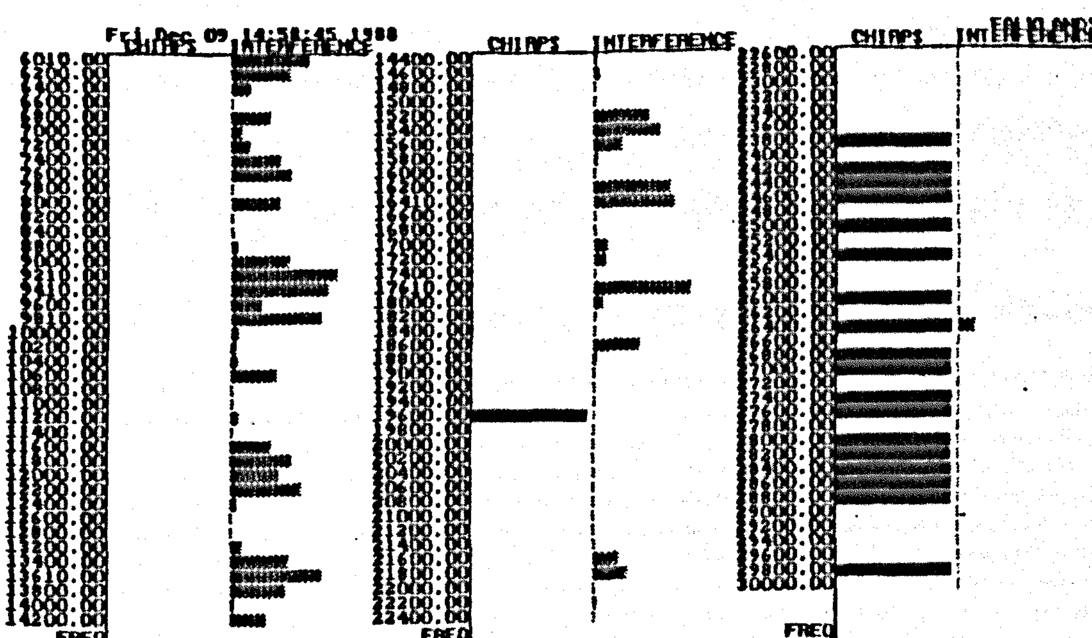


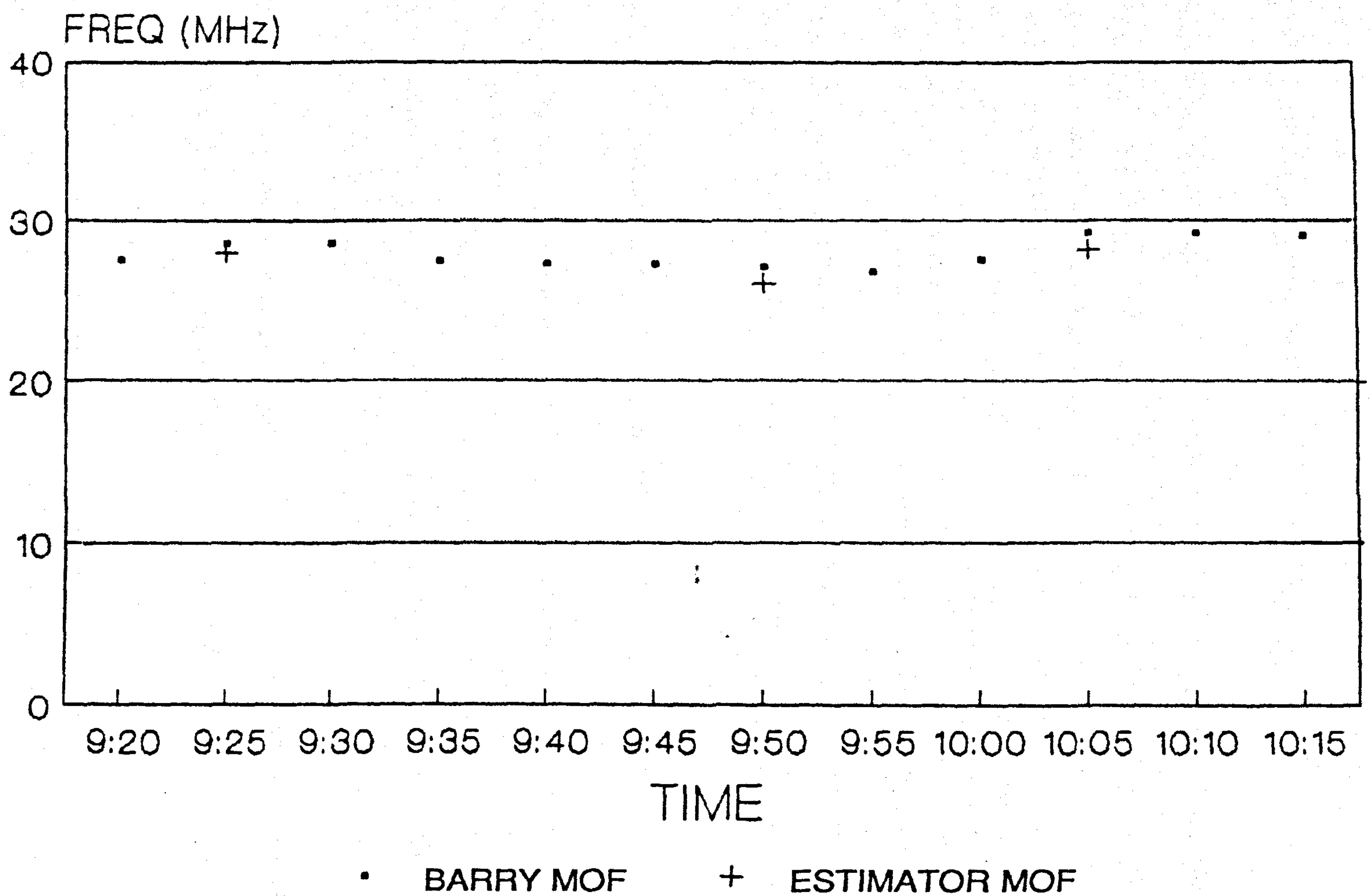
FIGURE 7.18. PLOTS OF MUF DERIVED FROM

THE BARRY CHIRPSOUNDER RECEIVER

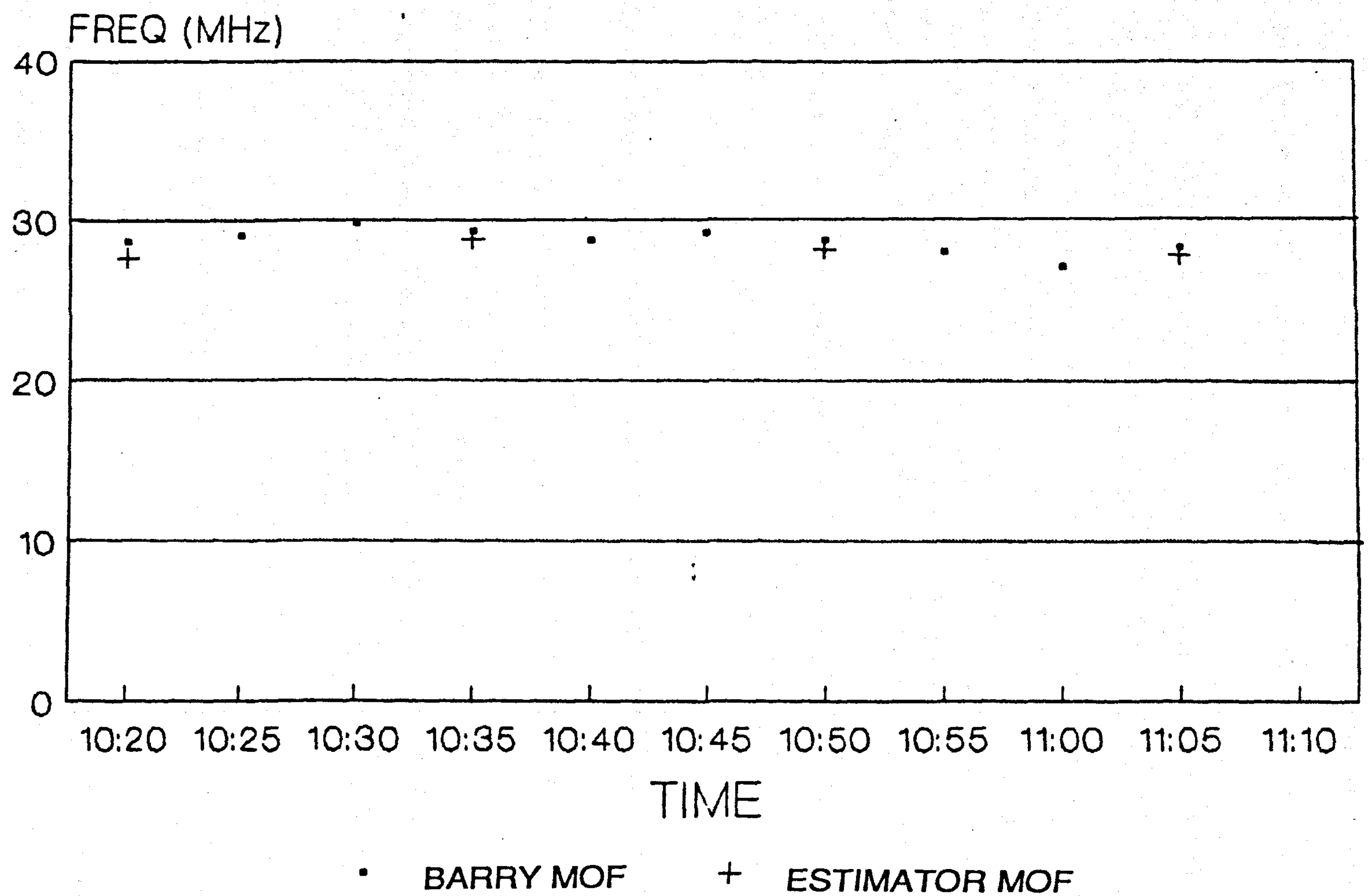
AND

THE PROPAGATION ESTIMATOR

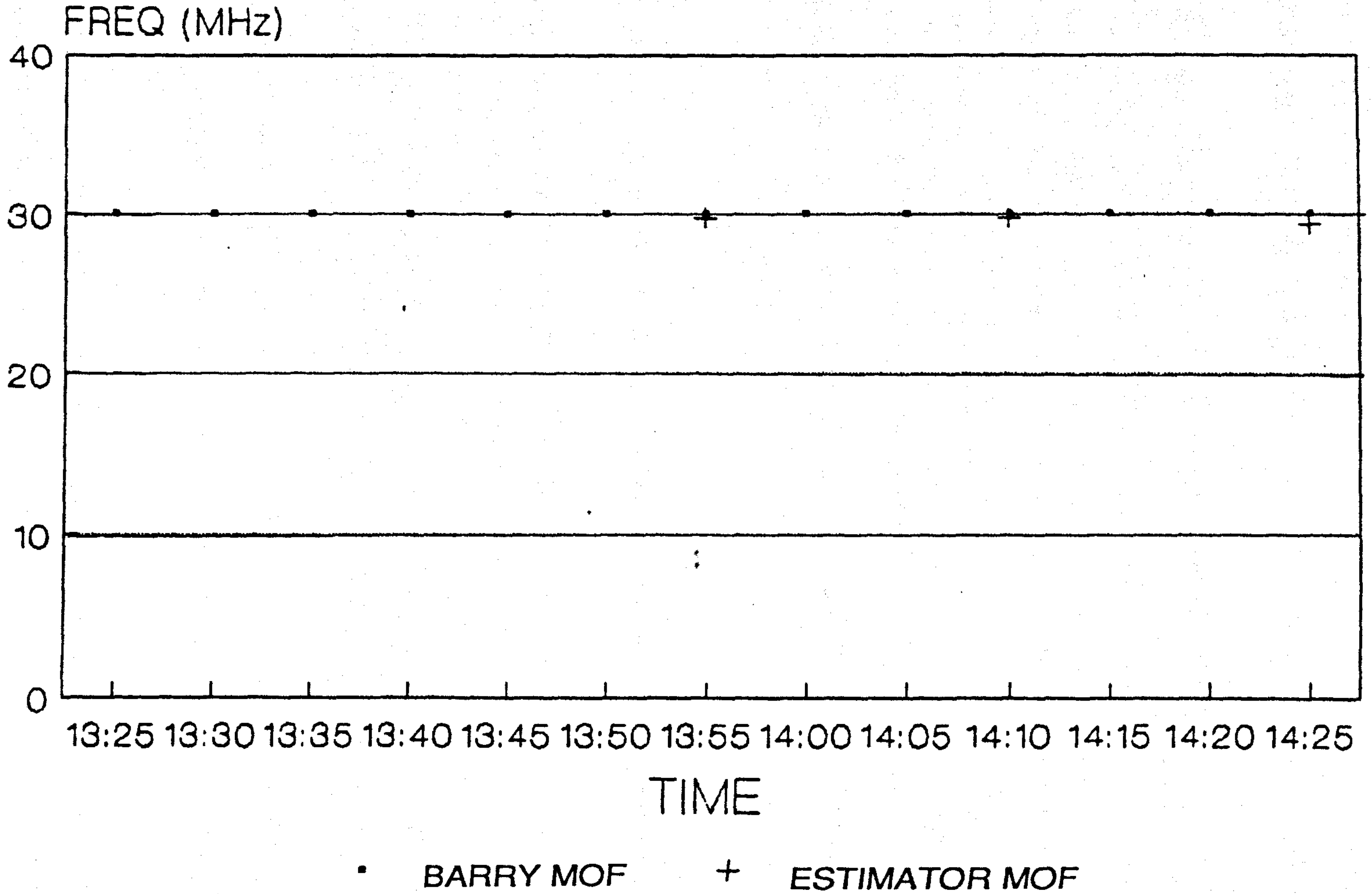
COMPARATIVE CHIRP TRIALS
BODO TO FARNBOROUGH (1)

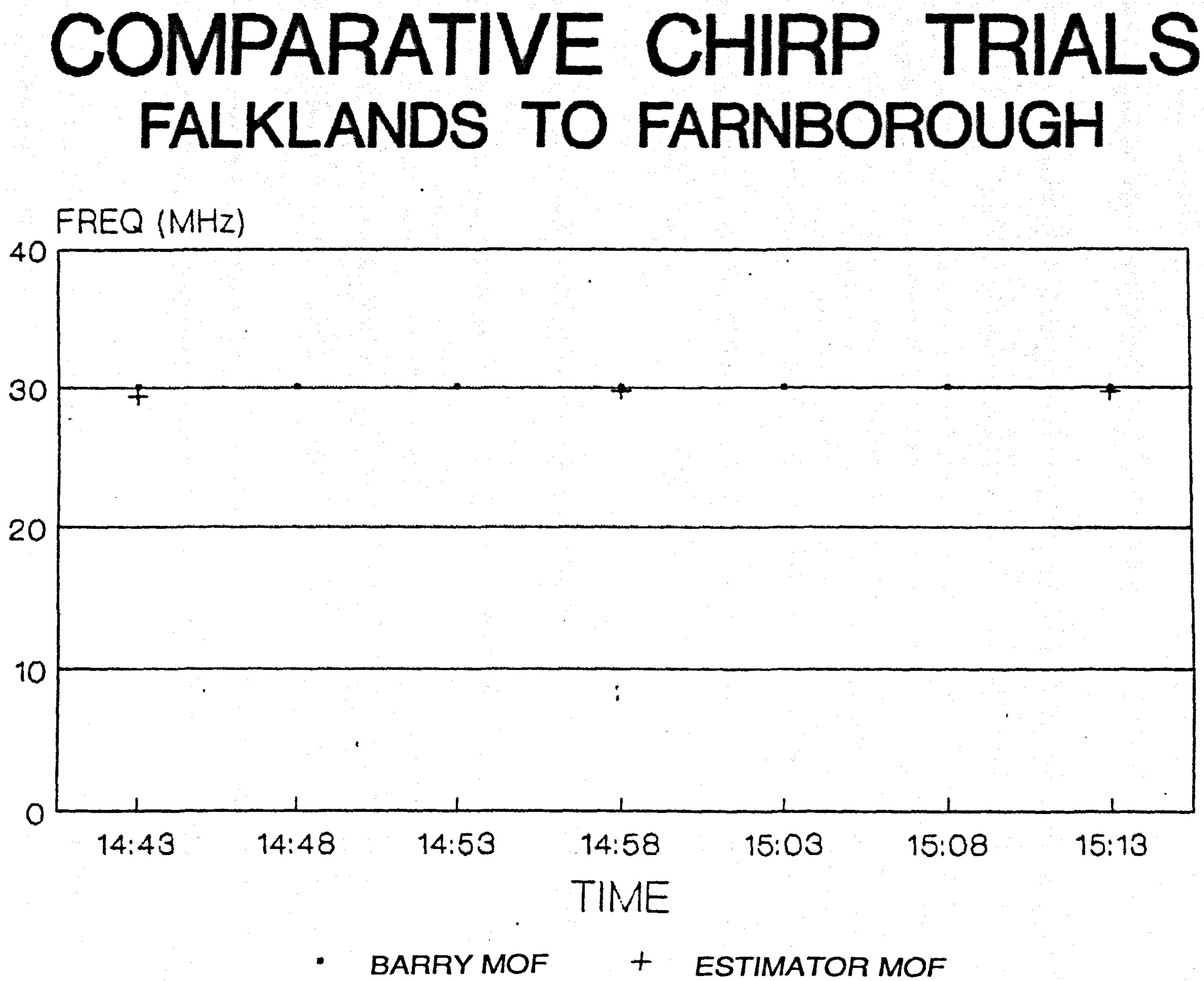


COMPARATIVE CHIRP TRIALS
BODO TO FARNBOROUGH (2)



COMPARATIVE CHIRP TRIALS
CYPRUS TO FARNBOROUGH




COMPARATIVE CHIRP TRIALS
FALKLANDS TO FARNBOROUGH
FREQ (MHz)
40
30
20
10
0
14:43
14:48
14:53
14:58
15:03
15:08
15:13
TIME
BARRY MOF
+ ESTIMATOR MOF

COMPARATIVE CHIRP TRIALS
MILLTOWN TO FARNBOROUGH

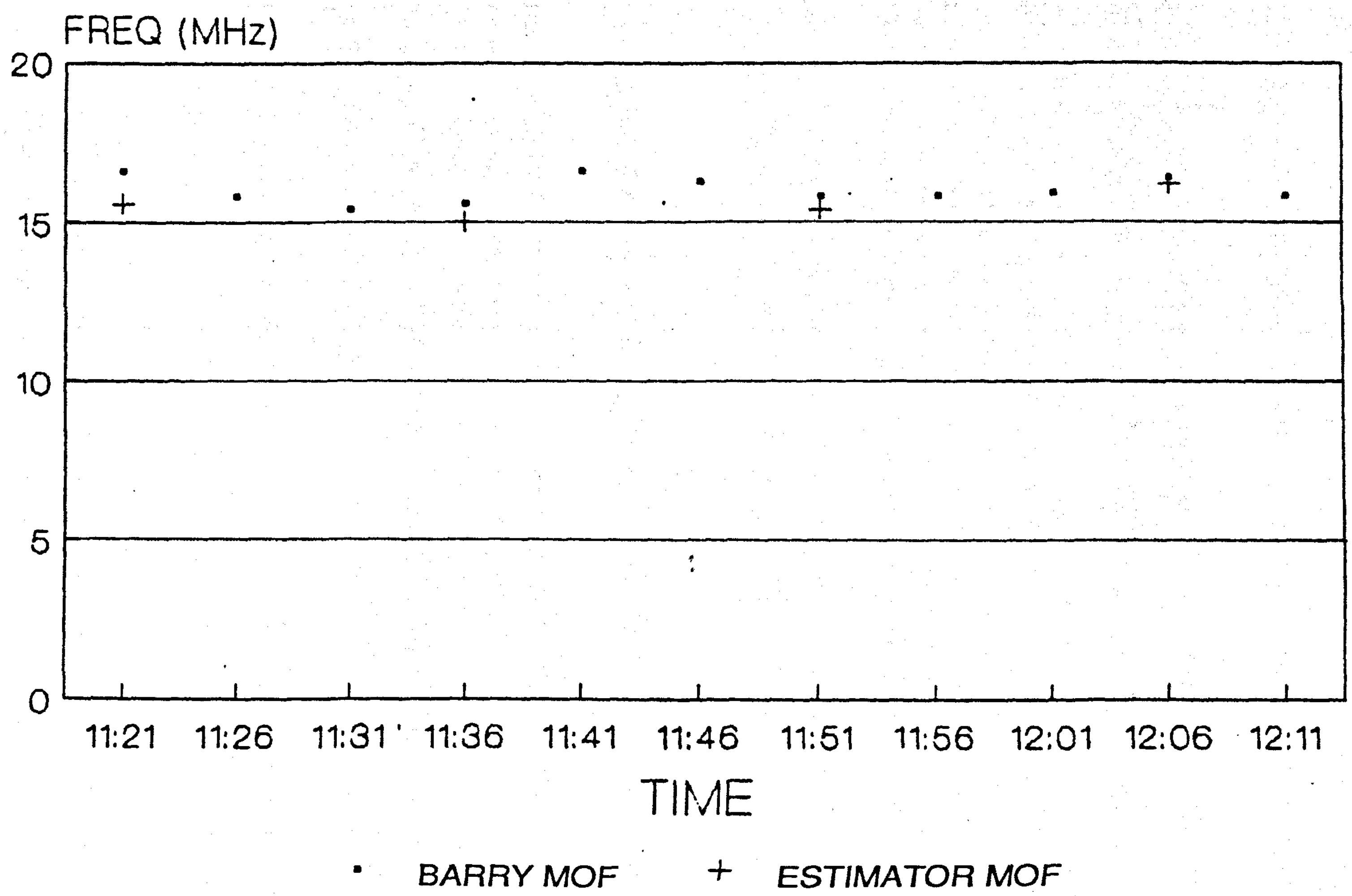
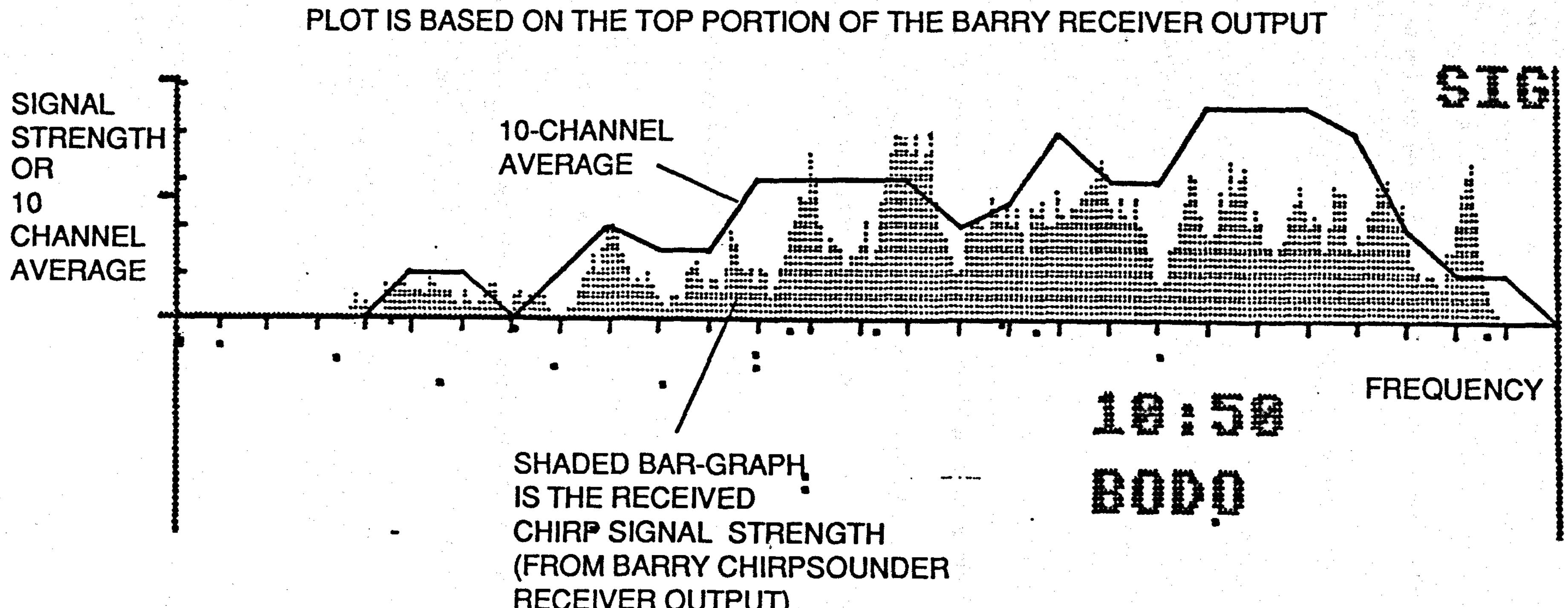


FIGURE 7.19. PLOT OF THE AVERAGE NUMBER OF CHIRP OCCURENCES IN A TEN CHANNEL SPAN (FROM THE PROPAGATION ESTIMATOR) AND THE RECEIVED CHIRP SIGNAL STRENGTH (FROM THE BARRY RECEIVER)



Further processing was performed on the results. An average of the chirp occurrences in a 10-channel span was plotted against the signal strength bar display on the top of the Barry output. This bar-graph represents the power of the chirp plus noise minus the noise power on its own, thus giving an indication of the strength of the received chirp. A reasonable degree of correlation was found between the two plots, as can be seen from the example shown in Figure 7.19. 3-and 5-channel span averages were also compared with the chirp strength graph, but the degree of correlation was less. It is therefore possible to produce a rough plot of the received chirp strength as a function of frequency, in addition to the MOF and LOF figures from the system. This could then be used to give an indication of the overall path attenuation on a channel, assuming that the RF elements of the system can be calibrated.

7.10 Conclusions and Potential Enhancements

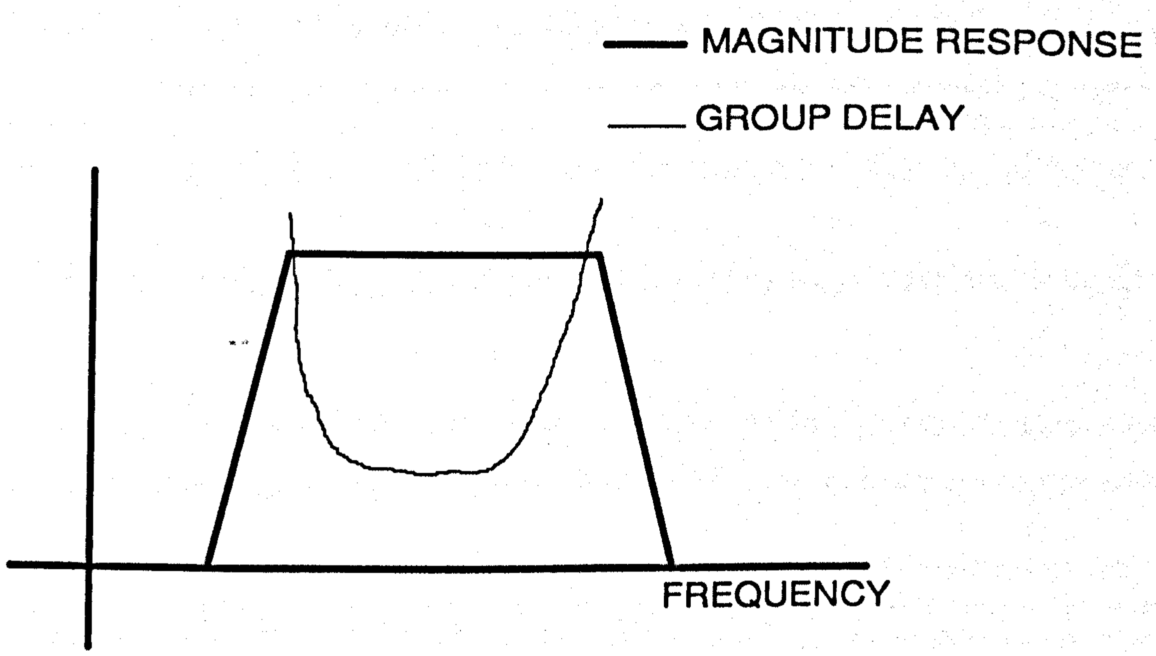
A chirp-based propagation estimation system has been designed, implemented and tested. This will be used in propagation model accuracy enhancement algorithms by a future research programme.

It is felt that the propagation estimator and chirp detector could be improved in the following areas:

i) **Filter Taps.**

In order to extract more information about the ionosphere, such as the relative path attenuations of a system's frequency allocations, it is necessary to have analogue samples of the chirp waveform as filter taps – rather than the limited values that have been used.

FIGURE 7.20. GROUP DELAY CHARACTERISTICS OF BAND-PASS FILTERS



ii) **Receiver Bandwidth.**

A wider bandwidth receiver, and hence a larger matched filter structure, would improve the selectivity of the chirp detector. This would also aid the identification of the ionospheric path's characteristic features such as its multi-path profile. This extra information could be used to construct a "rough" ionogram, in the manner indicated in Figure 7.6.

iii) **Characterisation Of The Receiver.**

In general, band-pass filters, such as the filters used in the IF stage of a radio receiver, can cause severe phase distortion of signals, particularly at the edges of the pass-band. This is illustrated by the generalised characteristic shown by Figure 7.20. Such distortion would affect the performance of the chirp matched filter. Characterisation of the receiver IF bandwidth, which could be achieved by sampling the audio response of the receiver to an RF input chirp and then using these samples as filter taps, would thus enhance the chirp matched filter performance since allowance could then be made in the design for group delay variation.

iv) **Characterisation Of The Effects Of Interference.**

The results of the comparative trials have shown that certain types and levels of interference reduce the effectiveness of the chirp detector. If the nature of different signals and their levels could be assessed then steps could be taken to compensate for them. Thus the accuracy of the propagation estimator would be improved.

8 TEMPLATE CORRELATION

8.1 Introduction

It is the task of the HF frequency management system (FMS), embedded within the overall communications system architecture, to decide which, out of all the system's frequency allocations, is optimum at any given time. In order to achieve this goal the FMS requires detailed and accurate data concerning the propagation, natural noise and interference conditions prevailing on each allocation.

In a contemporary automatic HF communication system, such as that developed by Plessey Research (see Chapter 5), the FMS and the overall system operate in two phases: in the "pre-call" phase, a prediction of likely channel performance parameters is made before traffic is passed; in the "in-call" phase, frequency management information is extracted from either the communications traffic or probing signals.

During the pre-call phase, the propagation information is usually supplied by a propagation prediction program, which is normally based upon a standard algorithm, such as that described by (CCIR, 1986a). Data concerning the natural noise levels on a channel, for a particular site, can be obtained from a computer-based version of (CCIR, 1986b) and information regarding man-made noise levels is availablein (CCIR 1986c). Hence the FMS has available to it detailed sources of propagation and noise information.

There are no world-wide interference prediction models in existence biassed towards the requirements of automatic HF communication systems. Thus, there is currently no detailed input of interference information to automatic FMS's during the pre-call phase. In order to try to alleviate this problem, the Plessey HF system passively tests the level of noise and interference in the channels which have been labelled by the propagation model as having the ability to propagate. These noise/interference levels are then used as the interference input to the FMS. However, in the Plessey system there is no way of assessing the in-band structure of the interference, or the likely time variability of a channel's interference profile. Such information would provide a more comprehensive indication of the transmission capabilities of the system's allocations and would also enable modulation formats to be adapted in response to interference characteristics.

In the in-call phase, the combined effects of the propagation, natural noise, and interference conditions on a channel are assessed by real-time channel evaluation (RTCE). A "good quality" channel should thus be rated highly by the RTCE routines. Again, the structure of the in-band interference is not assessed nor is its variability with respect to time. The channels are ranked in order of throughput, the best channel being the one with the highest rank.

The provision of interference information for an FMS is especially important in spectrally-congested regions, such as Western Europe, where interference can be the factor limiting HF communication system performance (Gott, 1983). Attempts have been made to model interference characteristics, but such models are still in the early stages of their development. One such model is described in (Laycock et al, 1988), the details of which are given in Section 3.5.

It is not a general model in the sense that it cannot produce interference information for all HF paths. The spectral occupancy results produced by it are valid only for a limited geographical area. Also, it is difficult to see how this model could be applied to channel assessment in an automatic FMS. An extensive, world-wide interference data gathering and data processing exercise would be necessary to produce a more general and useful model.

Hence the designers of automatic HF communication systems are unlikely to have available to them a systems-biassed model of interference within the time-span of the next sun-spot cycle (this represents the minimum length of time required to log enough results to produce a global interference model in the manner described by (Laycock et al, 1988)). FMS's constructed during this time must rely for their interference input on measurements made on frequency allocations during message transmission, and during times when the system is passive. Such measurements could also be logged continually and the results processed in order to try to predict trends in the usage of channels. This would enable the user's own transmissions to avoid interference more often.

Template correlation is a deterministic technique which aims to provide detailed information about the interference conditions on HF channels, both during the pre and in-call phases of the operation of the overall comunications system. It enables the FMS to select optimum channels and the optimum modulation scheme for each channel, with reference to the in-band interference structure. It also provides the FMS with the ability to determine whether or not a particular channel can satisfy the user's particular transmission requirements, eg required signal-to-noise ratio, bit-rate etc.

8.2 The Theory of Template Correlation

It can be shown that for maximum information rate in a noisy communications channel, the power spectrum of the desired signal should be adjusted so that the sum of the signal and noise power spectral densities is frequency-independent (Goldman, 1953). This condition is represented by Figure 8.1.

Referring to Figure 8.1:

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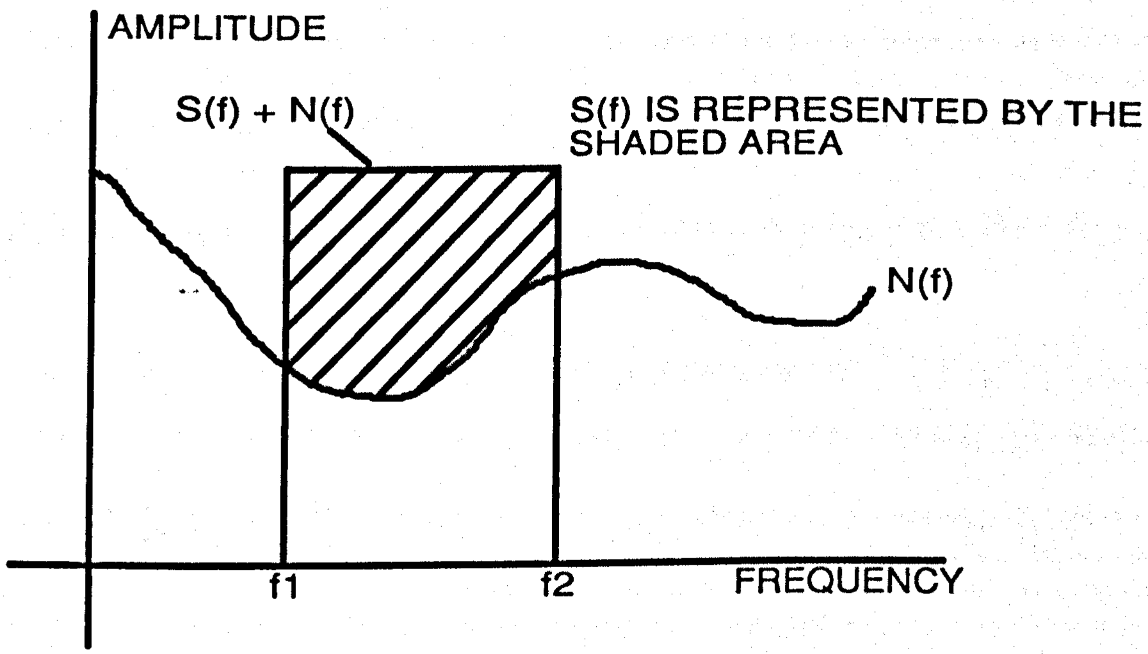
Referring to Figure 8.1:

FIGURE 8.1. SPECTRAL SHAPING REQUIREMENTS FOR MAXIMUM INFORMATION RATE



$$\int_{f_1}^{f_2} S(f).df = P \tag{8.1}$$

$$\int_{f_1}^{f_2} N(f).df = N \tag{8.2}$$

where $S(f)$ = power spectral density of the signal.

$N(f)$ = power spectral density of the noise.

P = total signal power in the range f_1 to f_2.

N = total noise power in the range f_1 to f_2.

Assume that the noise is Gaussian. Assume also that there is also no intersymbol influence in the frequency domain and that the signal has Gaussian properties.

Consider the frequency band f_1 to f_2 to be divided into smaller bands so that $N(f)$ is approximately constant in each.

Now, the entropy of any continuous distribution is given by

$$H = -\int_{-\infty}^{\infty} p(x).\log_2(p(x)).dx \tag{8.3}$$

and that the probability density function $p(x)$ for a variable x is given by:

$$p(x) = \exp(-x^2/2\sigma^2) \;/\; (2\pi\sigma^2)^{0.5} \tag{8.4}$$

where σ is the standard deviation of the Gaussian distribution.

By taking anti-logs and performing the entropy integral, the result below is obtained:

$$H = \log_2(2\pi e\sigma^2)^{0.5} \tag{8.5}$$

So for a Gaussian source, sampled at n points, the entropy of each point h is given by :

$$h \quad = \quad \log_2(2\pi e\sigma_I{}^2)^{0.5} \qquad (8.6)$$

where σ_I is the root mean square value of x about the mean at the sample point (as the mean square of x about the mean corresponds to the variance σ^2).

For our noise source, let $N = \sigma_I{}^2$, ie let the mean square noise power in a 1Ω load be the variance of the distribution (as per definition of variance).

Thus the entropy of each sample point is:

$$h \quad = \quad \log_2(2\pi eN)^{0.5} \qquad (8.7)$$

This is the entropy per degree of freedom per sample. If Nyquist sampling is assumed and the number of degrees of freedom is taken as one, as sampling is carried out in either frequency or time, then the total entropy is given by :

$$H \quad = \quad T.W.\log_2(2\pi eN) \qquad (8.8)$$

where: T = the duration of the signal.

W = the bandwidth of the signal.

Hence, the entropy of the combined signal and noise in a narrow bandwidth Δf is:

$$H(s+n) \quad = \quad T.\Delta f.\log_2(2\pi e(S(f).\Delta f + N(f).\Delta f)) \qquad (8.9)$$

The entropy of the noise alone in the bandwidth Δf is:

$$H(n) = T.\Delta f.\log_2(2\pi e(N(f).\Delta f)) \qquad (8.10)$$

The information rate, R, is then given by:

$$R = (H(s+n) - H(n)) / T \qquad (8.11)$$

as $T \rightarrow \infty$

Thus, the information rate in bandwidth Δf is:

$$R(\Delta f) = \Delta f.\log_2((S(f) + N(f)) / N(f)) \qquad (8.12)$$

Hence, the information rate in the whole bandwidth f_1 to f_2 is given by integrating equation 8.12 over this range, as follows:

$$R(W) = \int_{f_1}^{f_2} \log((S(f) + N(f)) / N(f)).df \qquad (8.13)$$

It is required to maximise this rate subject to the following constraint:

$$P = \int_{f_1}^{f_2} S(f).df \qquad (8.14)$$

This is achieved via the calculus of variations, which yields the result:

$$(\delta/\delta S(f)) \log(1 + (S(f)/N(f))) + \lambda = 0 \qquad (8.15)$$

where λ = a constant.

Therefore:

$$N(f) /(S(f) + N(f)) \times 1/N(f) = -\lambda \qquad (8.16)$$

Which gives the result:

$$S(f) + N(f) = -1/\lambda \tag{8.17}$$

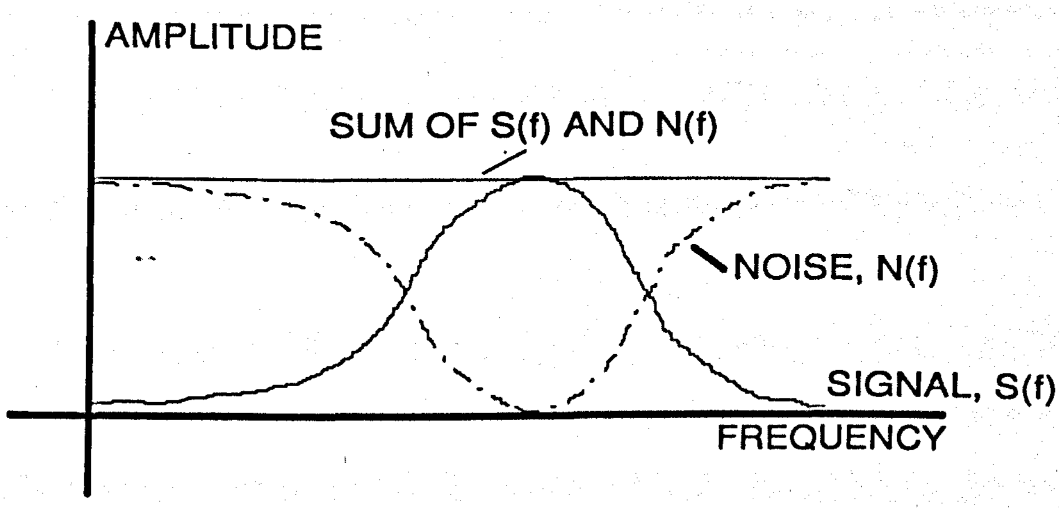
ie for maximum information rate, the sum of the noise and signal power spectral densities should be frequency independent (as shown in Figure 8.1).

At points in the bandwidth where the noise power exceeds maximum signal power, then obviously the condition described by equation 8.17 cannot be met. In this case, signal power is not wasted in trying to overcome the noise peaks and no signal power is applied in such regions.

The above ideal condition is valid for Gaussian noise conditions. However, HF channel noise is typically non-gaussian and thus it is probably more valid to consider the meaning of equation 8.17 in general terms, ie the maximum information rate condition described above is achieved when most of the signal power is placed where the noise power is least. For a practical communications system, the degree of spectral shaping of the signal that is required to meet the ideal condition of equation 8.17 is impossible to achieve with commercially available modems. However, it is possible to achieve an approximation to the ideal case via, for example, a computer-controllable, frequency-agile FSK modem, which has the ability to position modulator tones anywhere within the channel bandwidth. Modems of this type are now becoming more readily available.

Template correlation is a technique which allows any practical communications system to adjust its modulator output so that the information rate in the channel is maximised via an approximation to equation 8.17. The mechanism by which this is achieved is described below.

FIGURE 8.2. IDEAL SIGNAL POSITIONING WITHIN A NOISY CHANNEL



8.3 Principles of Template Correlation

8.3.1 The Basic Algorithm

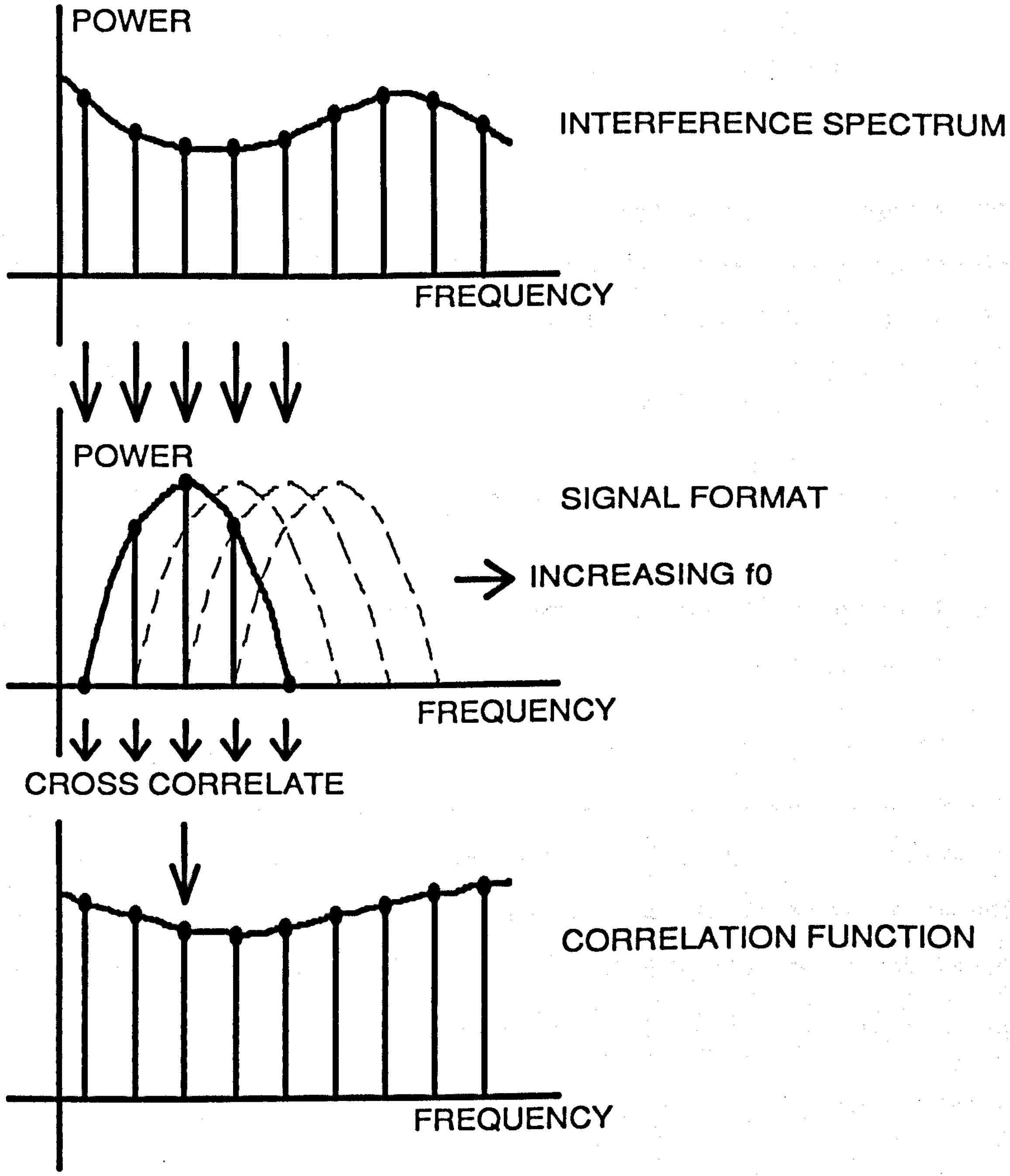
In order to maximise the information transfer rate over a noisy channel, it has been shown that the condition described by equation 8.17 must be met. In a practical communications system, where only limited spectral shaping of the signal format is feasible, an approximation to this can be made by placing most of the signal power where there is the least amount of noise power.

Thus, for an automatic communications system, there is the problem of establishing those positions within the channel bandwidth, for a particular modulation format, that will enable the maximum information rate condition to be achieved. Template correlation allows these positions to be located automatically.

Consider the situation shown by Figure 8.2: this diagram depicts the ideal transmission position for a signal tone in a noisy channel, ie the signal power is greatest where the noise power is least. In this case, the tone has been placed where the noise spectrum is the inverse of the signal spectrum.

Now consider the situation where it is required to find the ideal position for a particular signal format within a noisy channel: if a frequency domain cross-correlation is performed between the signal and the interference spectra, over the whole of the channel bandwidth, then the points of greatest similarity will appear as maxima in the correlation function. Conversely, points of least similarity will appear as minima. The principal minimum will be where the maximum information rate condition shown in Figure 8.2 occurs.

FIGURE 8.3. TEMPLATE CORRELATION MECHANISM



Hence, by performing the above frequency domain cross-correlation and then locating minima in the resulting correlation function, the points within a noisy channel that are most suitable for use by a particular modulation format are found.

The availability of cheap, powerful digital signal processing (DSP) devices, such as the TMS 320, means that is feasible, from both economic and technical points of view, to automate template correlation and embed it within the terminals of an automatic HF communications system.

In a practical implementation of template correlation, in order to compute the required correlation function, it is first necessary to provide, in sampled form, a template of the wanted signal power density spectrum (PDS), S(f). The noise/interference PDS, N(f), in the channel bandwidth is also obtained in sampled form, eg via a fast Fourier transform (FFT). The template correlation algorithm is then effected by a cross-correlation between S(f) and N(f), as the signal profile is shifted by a variable frequency offset, f_o, with respect to the noise profile, as illustrated in Figure 8.3. This digital cross-correlation function is of the form:

$$\Phi_{ns}(f_o) = \int_B N(f)S(f + f_o) . \, df \qquad (8.18)$$

Where $\Phi_{ns}(f_o)$ takes a minimum value, ie N(f) and S(f) have greatest dissimilarity, the value of the frequency offset, f_o, identifies the position within the channel bandwidth where the signal, S(f), should be centred to maximise the transmission capacity. In communication terms, this position will give a minimum error rate at the receiver for the specified transmitted signal format.

FIGURE 8.4. THRESHOLDING OF SPECTRA

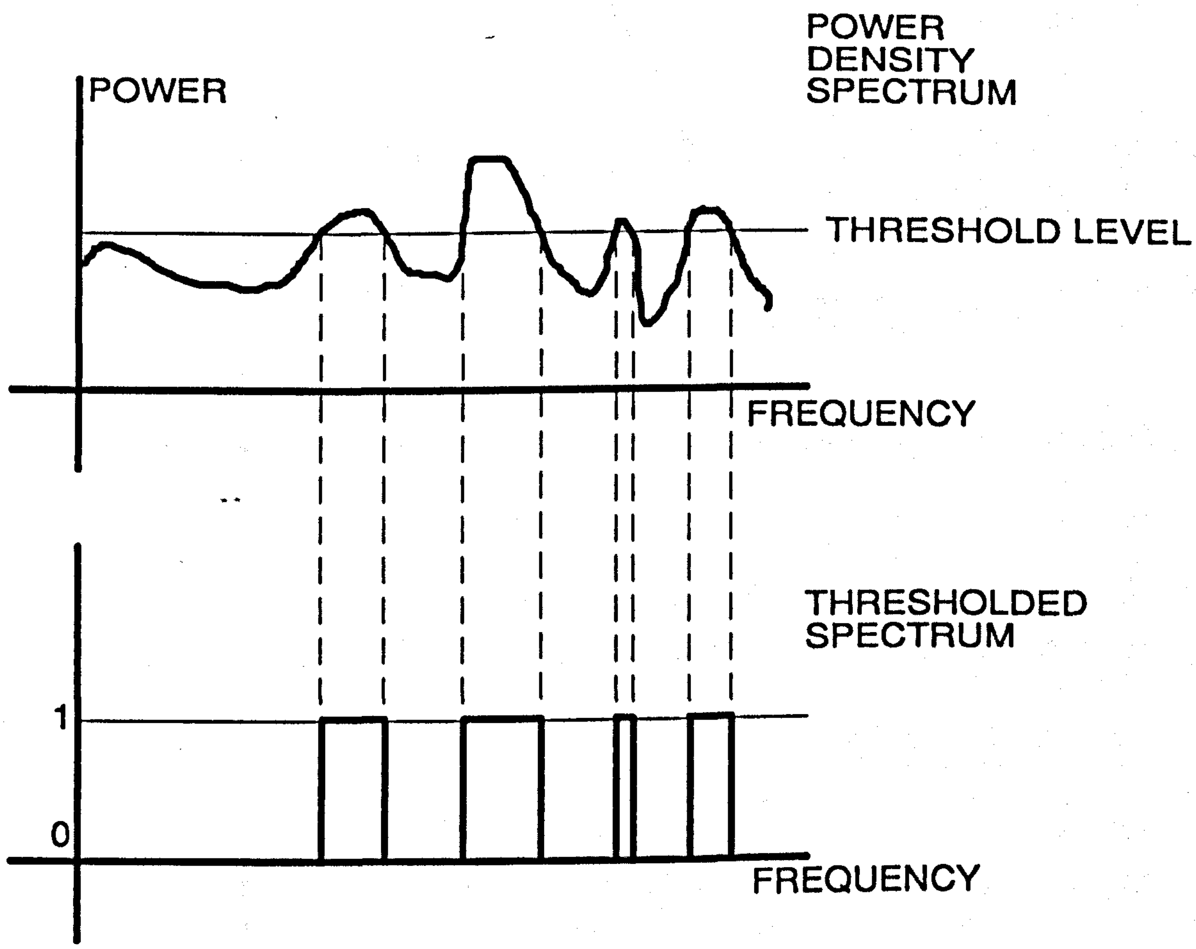


FIGURE 8.5a. AMPLITUDE SPECTRUM OF SINC(X) TONE

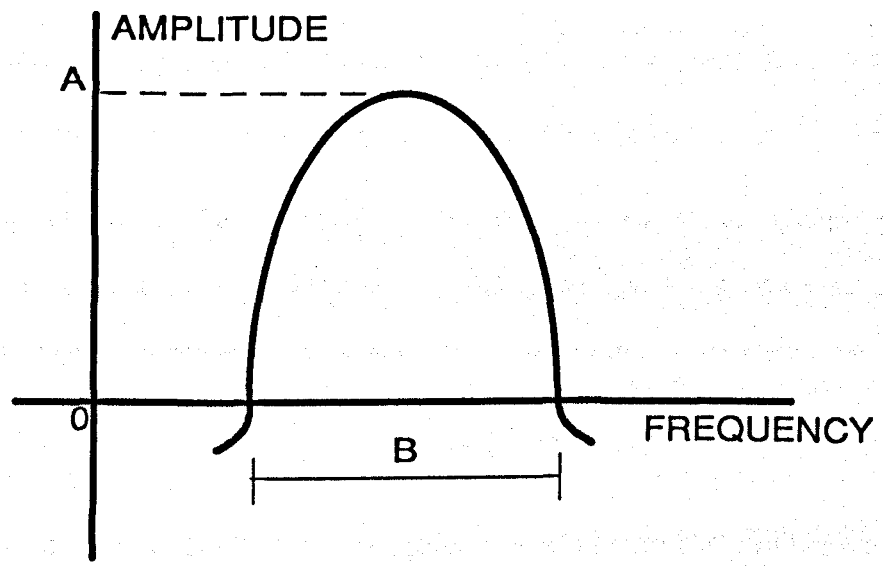
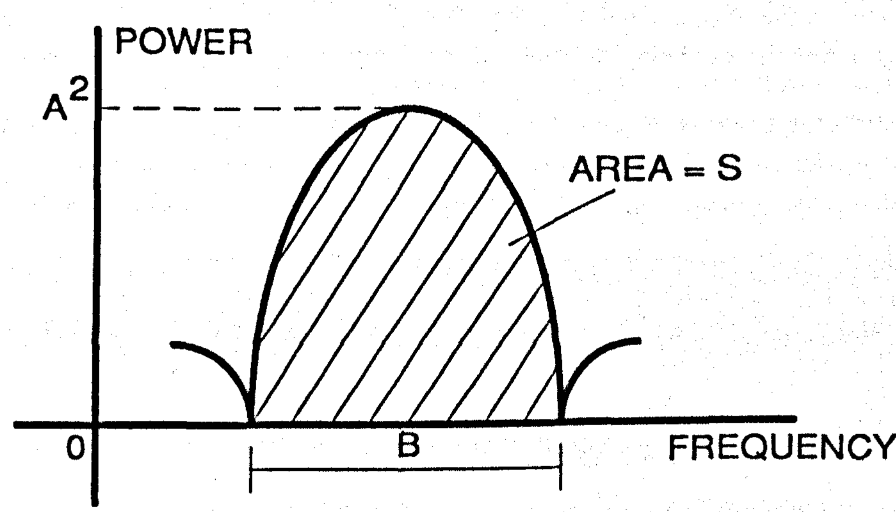


FIGURE 8.5b. POWER SPECTRUM OF SINC(X) TONE



8.3.2 A Modified Algorithm Incorporating Thresholding

In order to reduce the amount of processing required to perform template correlation, the basic algorithm can be simplified by the application of a technique known as "thresholding". This converts the two PDS's to be correlated into a binary form in the manner shown by Figure 8.4. Storage and processing time will thus be reduced if the noise/interference and signal spectra are represented in this fashion.

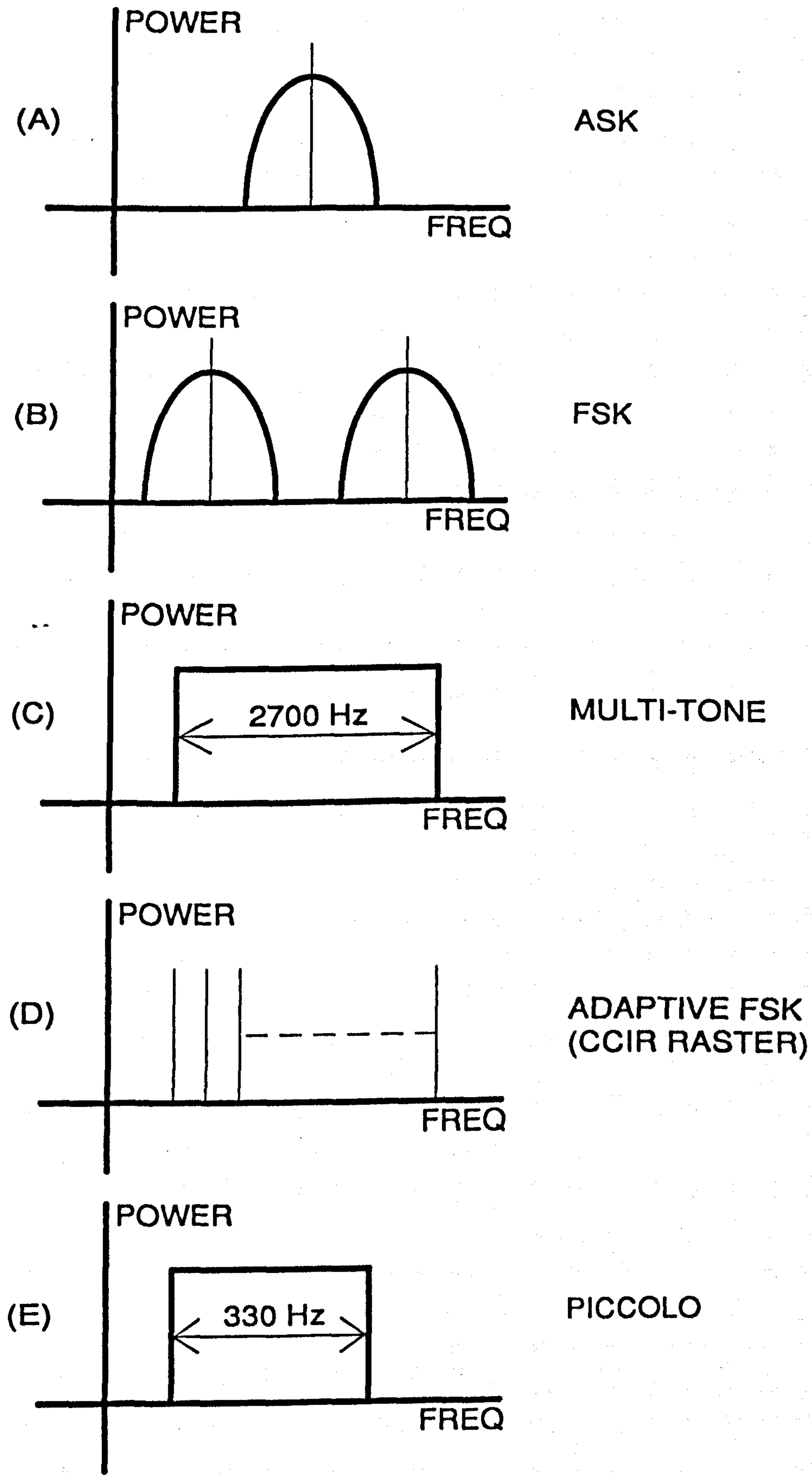
The level at which the threshold is set can be related to system performance parameters, such as the minimum required signal to noise ratio at the receiver and, for digital modulation schemes, bit error rate. Figure 8.5a shows one modulated tone resulting from binary frequency-shift keying (FSK) modulation (Schwartz, 1981). The tone has first null-to-first null bandwidth B and a peak amplitude of A. B is also the detection bandwidth of the corresponding matched filter. Figure 8.5b shows the PDS of the same signal and the shaded area in this diagram represents the total power of the signal tone in the detection bandwidth B. If one assumes that logic 1's and 0's are equi-probable, ie the source has maximum entropy, then the mean signal power in the bandwidth B will be half the shaded area in Figure 8.5b, ie S/2. The probability of error for matched filter detection of FSK in Gaussian White Noise, with zero multi-path conditions is given by (Schwartz, 1981):

$$P_e = 0.5.e^{-\psi/2} \qquad (8.19)$$

where ψ = (mean signal power) / (mean noise power) : both in the detection bandwidth B.

In a practical implementation of template correlation, the mean signal power could be determined at the receiver by performing a fast Fourier transform on the

FIGURE 8.6. TEMPLATE CORRELATION SIMULATOR MODULATION FORMATS



incoming signal and then calculating the area shaded in Figure 8.5b. This could be achieved by adding up the samples in the bandwidth B and then by using a look-up table to translate this area into a power value. The power figure is then used, along with the minimum required probability of error for the system, and equation 8.19, to determine the maximum allowable mean noise/interference power. This figure represents the level at which the noise/interference and modulation format spectra are limited to ensure that the mean signal to noise ratio is always greater than the minimum required value.

8.4 Computer Simulation

In order to further investigate the technique, a template correlation simulator was constructed. This was software-based, written in Pascal on an IBM pc XT.

For the purposes of the simulation, noise/interference and modulation spectra were sampled in the frequency domain at 10 Hz intervals. In this way, a single interference spectrum from a 3 kHz channel could be stored in 300 samples. The correlation mechanism used is as described in section 8.3.1.

The signal templates available for correlation are as shown by Figure 8.6 (a)-(e):

(a) amplitude-shift keying (ASK).

(b) frequency-shift keying (FSK).

(c) multiple-tone frequency-division multiplex modem.

(d) adaptive FSK in which the tone frequencies can be varied.

(e) PICCOLO format (Ralphs, 1985).

The noise/interference profiles fall into two categories: those which represent digitised versions of "real" HF channel spectra (supplied by Dr. G. Gott, UMIST)

FIGURE 8.7. TEMPLATE CORRELATION OUTPUT

FIGURE 8.7a RANDOM NOISE SPECTRUM AND PICCOLO SIGNAL

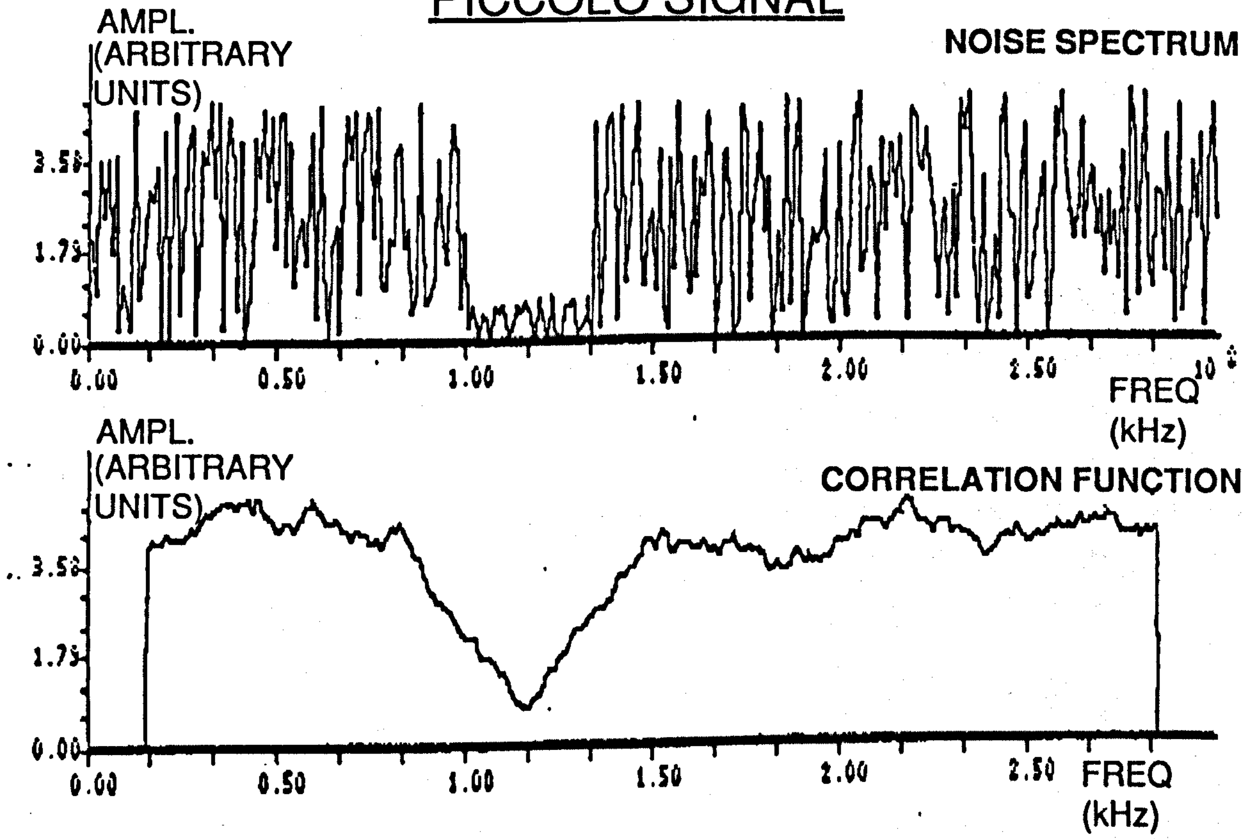


FIGURE 8.7b. "REAL" HF SPECTRA AND ASK SIGNAL

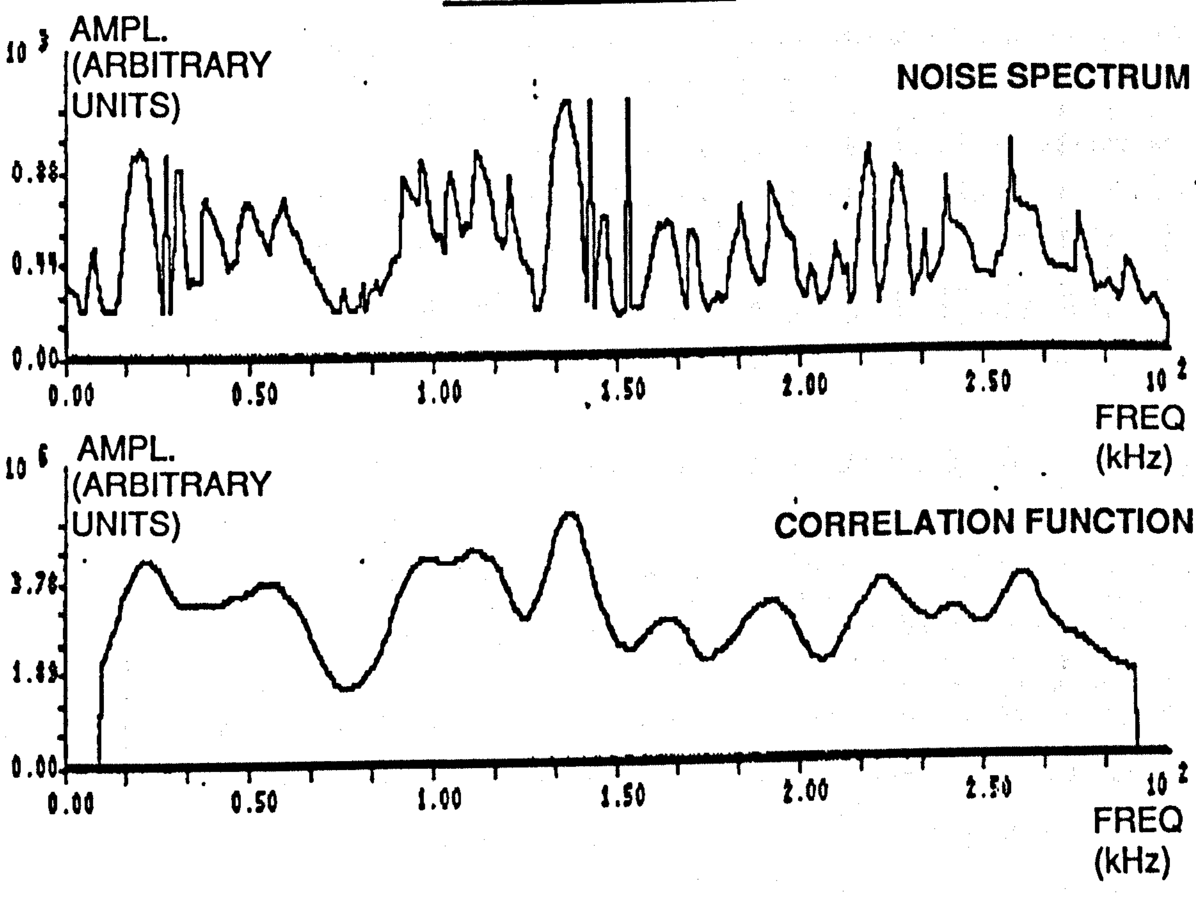


FIGURE 8.8. TEMPLATE CORRELATION OUTPUT

FIGURE 8.8a. RANDOM SPECTRUM (WITH GAPS) AND FSK SIGNAL

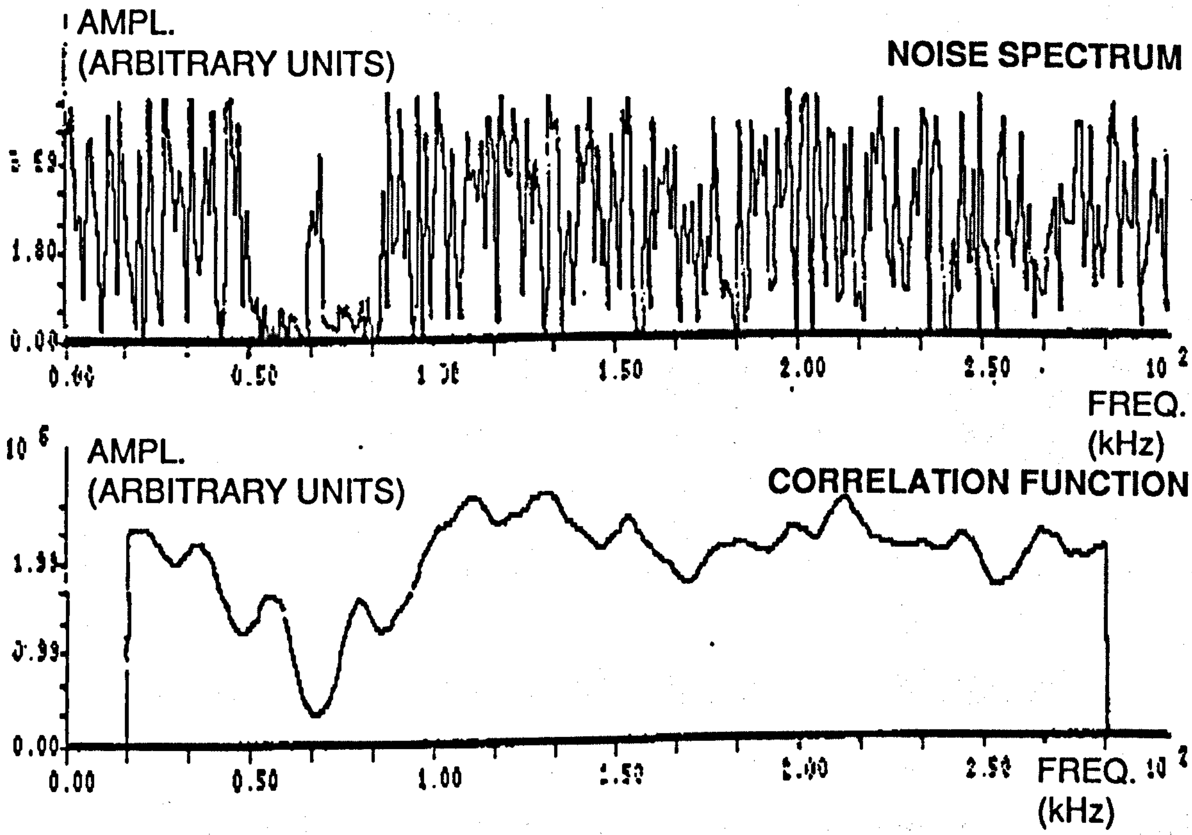
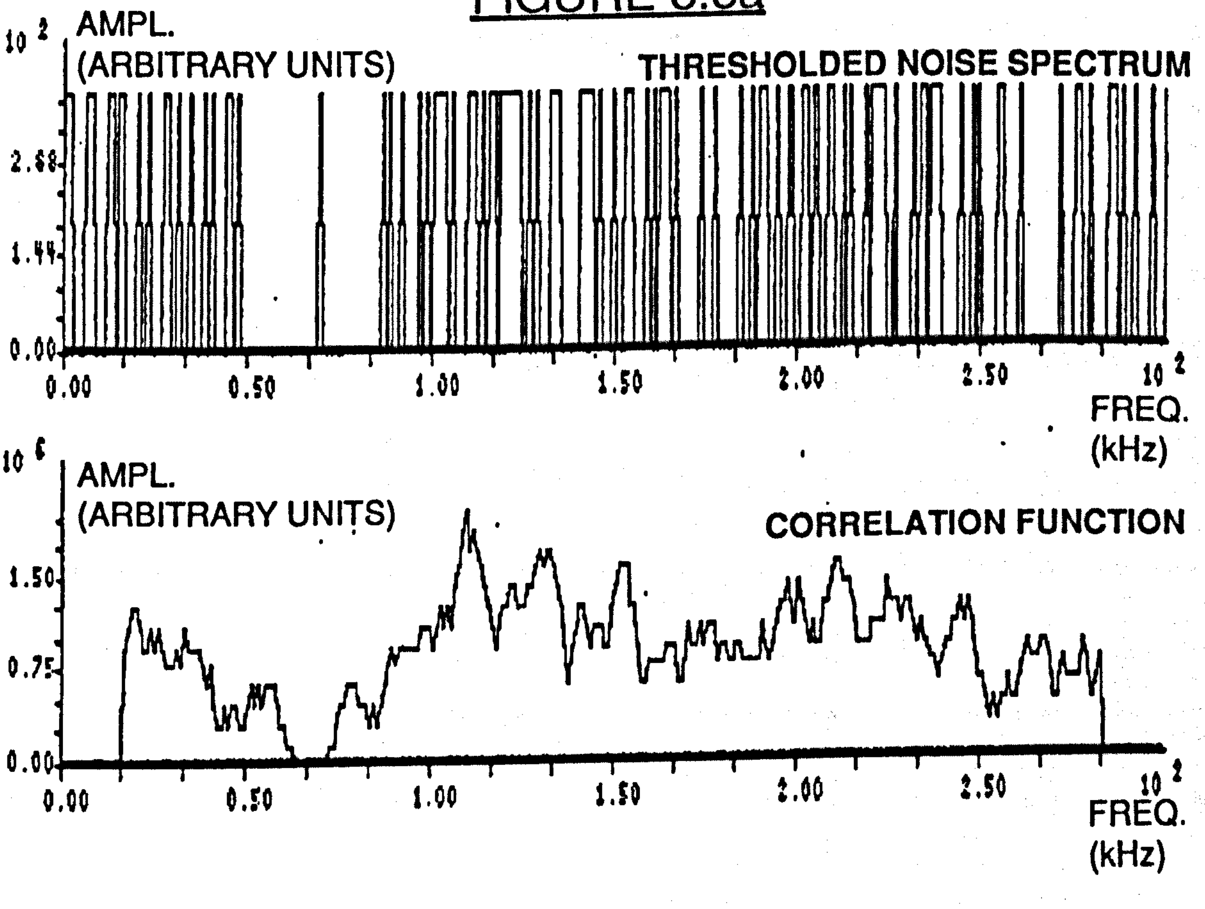


FIGURE 8.8b. THRESHOLDED VERSION OF FIGURE 8.8a



and so-called "random" spectra produced by the random number generator procedure provided by the Pascal programming environment used to develop the software. Noise/interference spectra can be edited by means of a "gap" generator which reduces the amplitude of the spectra over a user-defined frequency range. The thresholding algorithm described in Section 8.3.2 was also implemented.

Figure 8.7a shows the result of performing a simulated cross-correlation between a random-type spectrum, with a region of reduced amplitude "noise" 330 Hz wide, and a signal of type (e). The minimum correlation value indicating the optimum transmission position is clearly visible in the resulting correlation function. Figure 8.7b is more realistic in that the noise spectrum is a digitised version of measurements made on an HF channel. In this case, the signal template was of type (a) with a null-to-null bandwidth of 200 Hz.

Figure 8.8a shows a conventional correlation operation involving a signal of type (b) and a random number noise/interference spectrum. The same correlation has been carried out to yield the result in Figure 8.8b but, in this case, thresholding was applied. It can be seen from comparison of the two resulting correlation functions that the signal position selection decision would be the same in both cases, thus validating the thresholding simplification technique.

The results of the simulations carried out showed template correlation to be a useful technique for avoiding noise and interference in congested communications channels. However, it was felt that a more rigorous examination of the technique was necessary in the form of trials carried out on a laboratory bench.

8.5. Bench Trials of Template Correlation

8.5.1 Introduction

The aim of these trials was to further validate template correlation via tests carried out under more realistic conditions than those available on the simulator. In the simulated trials, template correlation was able to select transmission positions within the channel bandwidth which should provide the minimum error rate for a given signal format within that channel. However, the simulator did not allow this to be verified by the transmission of actual data over the noisy channel, with and without template correlation being used.

In an automatic HF communication system it would be the task of template correlation to provide the FMS with data on the prevailing noise and interference conditions in the system's frequency allocations. This information would then be used to adapt the signal format so that an approximation to the ideal minimum error rate condition, described by equation 8.17, could be made. From a communicator's viewpoint, this should also allow an increase in the overall throughput rate of the system.

Thus the bench trials were set up in order to ascertain whether template correlation could achieve one of its aims: to minimise the bit error rate on a channel for a given modulation format.

8.5.2 Trials Methodology and Apparatus

The apparatus used for the trials was developed over a period of one month, and the final configuration was as shown by Figure 8.9. The equipment used was as follows:

FIGURE 8.9. TEMPLATE CORRELATION BENCH TRIALS APPARATUS

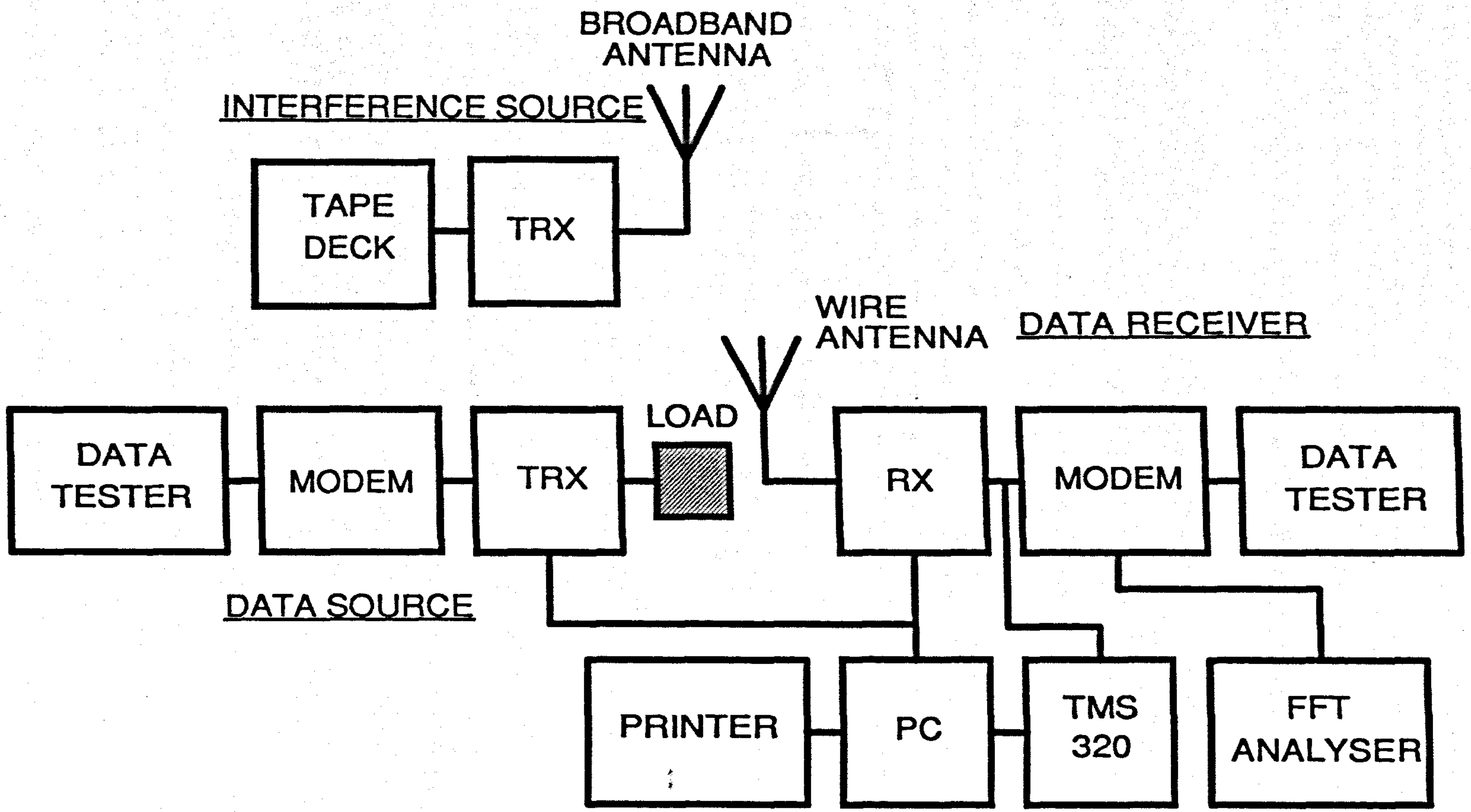
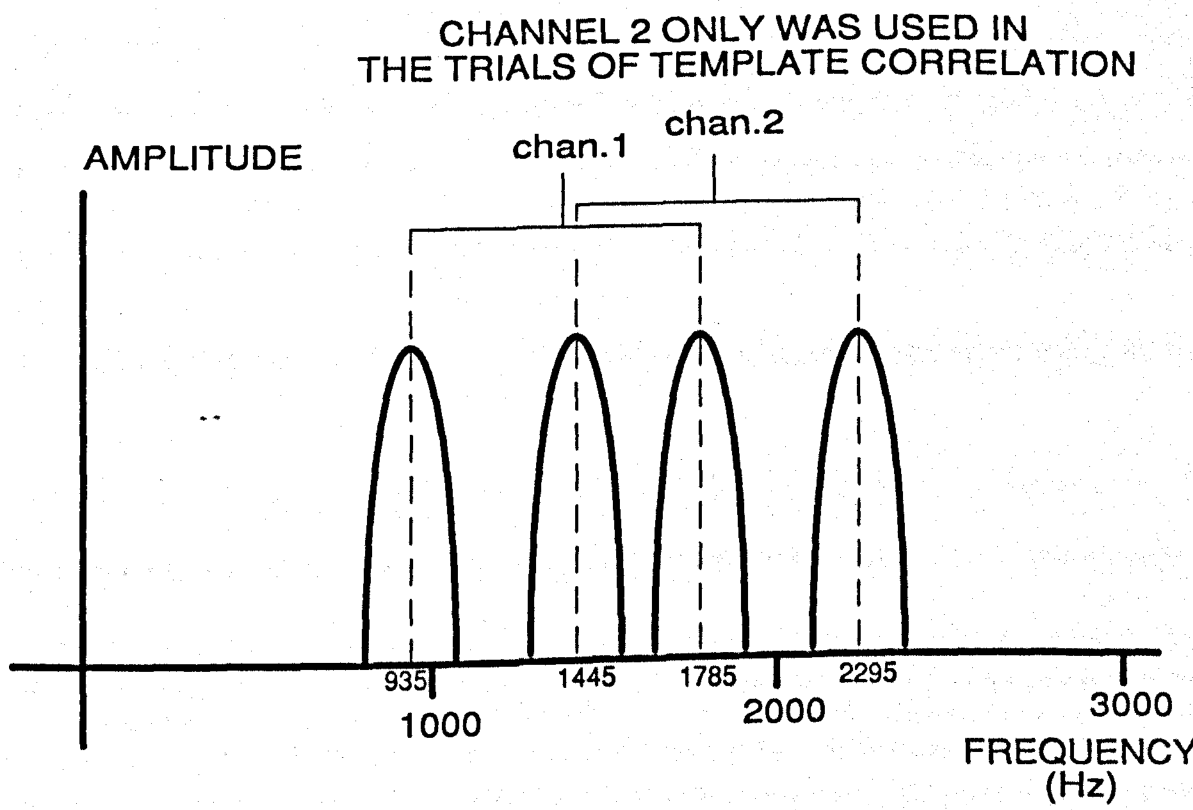


FIGURE 8.10. MODEM TONE POSITIONS



Data Testers.

Both of these were Trend DT-108 data testers. These units have the ability to generate messages (eg pseudo-random sequences) set the mode of operation (simplex or duplex), set the UART format and analyse errors in an incoming data stream. In the trials the DT-108's were configured to transmit and receive 63-bit pseudo-random sequences at a transmission rate of 150 bits/s. Parity checking was used and the receiver calculated the total number of blocks and bits received correctly (one block being a full 63-bit sequence) and also the number of blocks and bits received that were in error.

Modems.

Both of these were Cossor CGT 1092 HF frequency-exchange keying (FEK) modems. They each have two transparent data channels, each one capable of handling tranmsission rates in the range 50 to 200 bits/s. The modems were configured as in the Plessey HF system, ie the two tone pairs could be used in either a multiplexed or diversity configuration, giving overall data rates of 75, 150 or 300 bits/s. The aim of using these modems was to try and assess the benefits that template correlation would give to the existing Plessey system.

The tone positions within the channel bandwidth were as shown in Figure 8.10. These tone positions were set by switches on the modem circuit board. The demodulator uses switched-capacitor band-pass filters, with a fourth-order roll-off each side of the pass-band and automatic gain control (AGC) on a tone pair basis. In the trials carried out, the modems were configured for simple FSK using the second tone pair, at a data rate of 150 bits/s.

RF Equipment.

The transceivers used were Icom IC-735's and the receiver was an Icom IC-R71E. These are computer-controllable via a serial TTL interface, with the ability to set the frequency to the nearest 10 Hz. They were operated in upper sideband (USB) mode and the receiver had a bandwidth of 2.7 kHz at the 6 dB down points.

Tape Deck.

This was a RACAL Store-4 portable instrumentation recorder. It is a high quality tape recorder, capable of storing four hours of HF noise and interference, from a 3 kHz wide channel on one reel of tape.

Computing Equipment.

The pc was an IBM pc XT compatible (Victor VPC 11) equipped with a TMS 320C25 development card in one of its expansion slots.

FFT Analyser.

This was an Ono Sokki CF 910 dual channel FFT analyser.

The software written to implement the template correlation algorithm was coded in C for the PC and TMS 320C25 assembly language for the TMS board. The PC-based software was responsible for the following tasks:

i) Initialisation of the system, eg RF equipment interfaces and frequencies.

ii) Overall control of the TMS board, ie downloading of the FFT software from the PC to the board and the scheduling of its operation.

iii) The correlation of the channel interference spectrum (provided by the TMS board) and the modem spectral template.

iv) The location of an optimum transmission position within the channel bandwidth, ie a minimum in the correlation function.

v) The de-tuning of the data transmitter and receiver in order to effect the signal format frequency shift to avoid interference.

vi) The displaying of the results of the tests.

The software on the TMS 320 board performed a 128-point Cooley-Tukey fast Fourier transform (FFT). This gave a 64-point, positive frequency, power spectrum as its output. The results of 10 FFT's were averaged over a period of one second in an attempt to reduce the likelihood of an unrepresentative fading signal spectrum being passed on to the PC for correlation purposes.

In order to test whether the apparatus was functioning correctly, and to perform a simple test of template correlation, the tape deck was replaced by a sinewave generator. The output of the generator was adjusted so that its frequency was equal to each of the modem tones in turn. The transceivers were switched to transmit and the data source and sink were switched on. Using the FFT analyser for comparison of signals, the power output of the interference transmitter was adjusted so that the received data signal and sinewave interference were of comparable power levels. The numbers of block and bit errors were recorded for a single message of length 10^4 bits.

The data transceiver was then switched to "receive" and the template correlation software was run (the signal format template was the response of the demodulator filters to the 20 dB down points). After the software had adjusted the frequency of the data transmitter and receiver, the data transmitter was again switched to "transmit" and the data testers were switched on. The error characteristics were noted for the same message as that sent without template correlation. Thus, the overall throughput of the system could be assessed, with and without the template correlation algorithm, for identical channel interference conditions.

Trials were also carried out using an FSK modem (with the same output format as the data traffic) as the interference source. A recording of the output of the modulator of approximately 30 minutes duration, with a data tester providing the data input, was made and this was then used as the interference source.

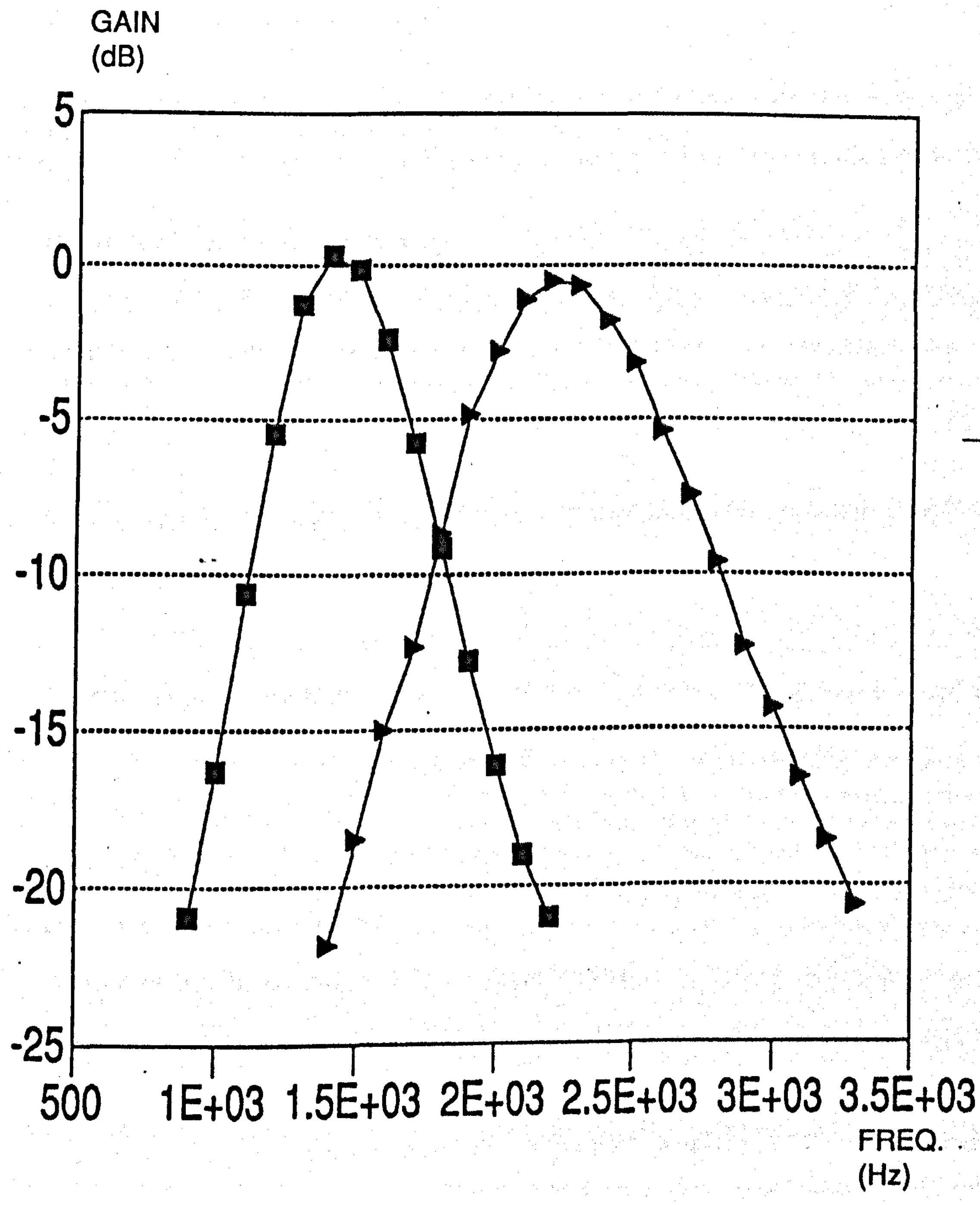
8.5.3 Results

The Cossor CGT 1092 demodulator proved to be unsuitable for template correlation trials. The demodulator band-pass filters have a fourth-order roll-off rate at each side of the pass-band. This meant that the template used for the initial trials occupied over half of the channel bandwidth, severely limiting the freedom of movement of the signal format within the band.

The demodulator also had AGC on a tone-pair basis, which could not be disabled without resorting to modification of the demodulator circuit boards. Thus the proximity of an interfering tone affected the amplitude of the signal tones at the output of the demodulator filters. Without detailed knowledge of the AGC scheme used by the demodulator it is impossible to select optimum in-band signal format transmission positions using template correlation. Hence another demodulator unit was sought which had a higher filter roll-off rate and which could be operated without any AGC.

One such FSK demodulator had been constructed for the use on another SERC-sponsored research project ("Improved Coding and Control for Communications over Dispersive Radio Channels with Non-Gaussian Noise", SERC Project Reference Number : GR/D/33557) by Mr. J. Hague, at the University of Hull. This unit also used switched-capacitor filter circuits to implement the demodulator filters; the roll-off rate was 48 dB / octave, ie eighth-order. The centre frequencies and filter bandwidths could be programmed via on-circuit board switches and the AGC circuitry could also be disabled.

FIGURE 8.11. AMPLITUDE RESPONSE OF DEMODULATOR FILTERS



The demodulator filters were programmed to correspond to the tone frequencies of the Cossor demodulator and the bandwidth of each tone filter was set to be 100 Hz. The resulting filter amplitude response is shown by Figure 8.11.

Template correlation trials were resumed with the new demodulator. A sample of the results obtained is shown below:

(i) **Sinewave interference on lower modem tone.**

Before template correlation was run:

Total number of blocks received	:	19
Number of blocks in error	:	19
Total number of bits received	:	1504
Number of bits in error	:	409
Bit error rate	:	27%

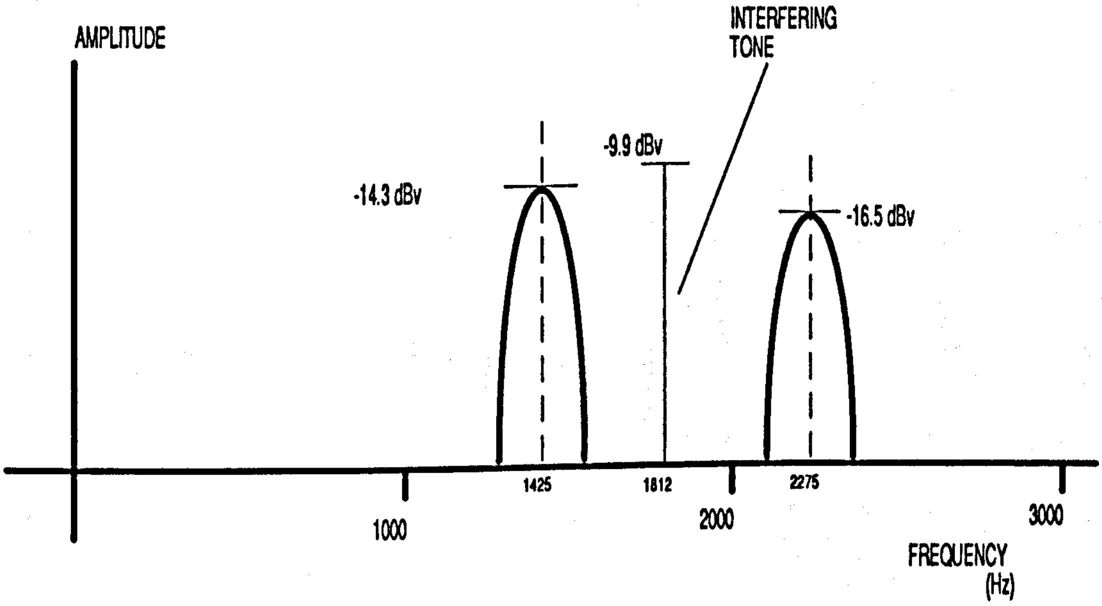
After template correlation was run:

Total number of blocks received	:	39
Number of blocks in error	:	31
Total number of bits received	:	3088
Number of bits in error	:	784
Bit error rate	:	25%

The output of the template correlation software and the FFT analyser was as shown by Figure 8.12a.

FIGURE 8.12a TEMPLATE CORRELATION BENCH TRIALS RESULTS

FFT ANALYSER OUTPUT



SOFTWARE OUTPUT

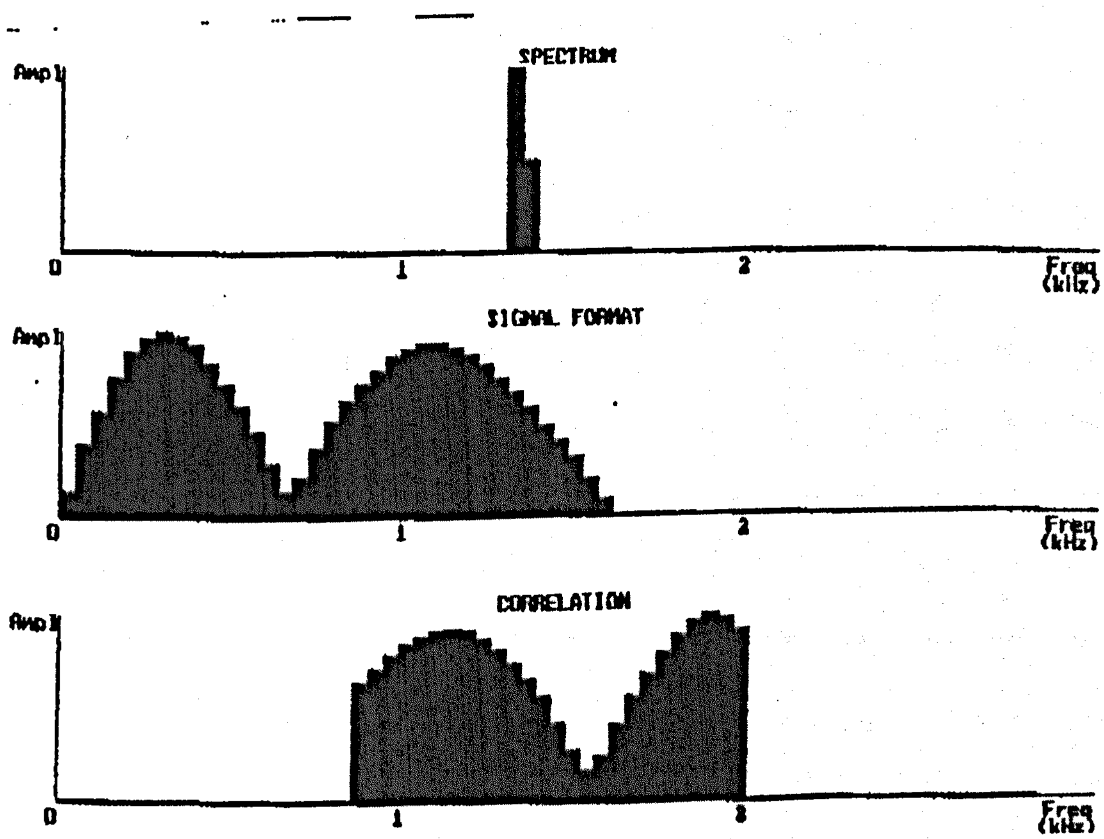
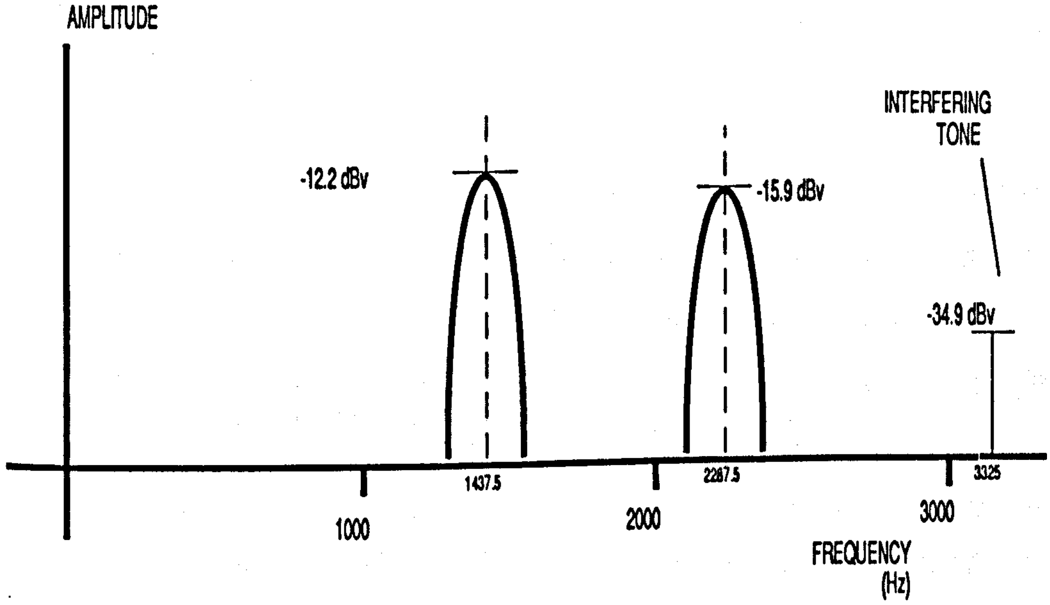
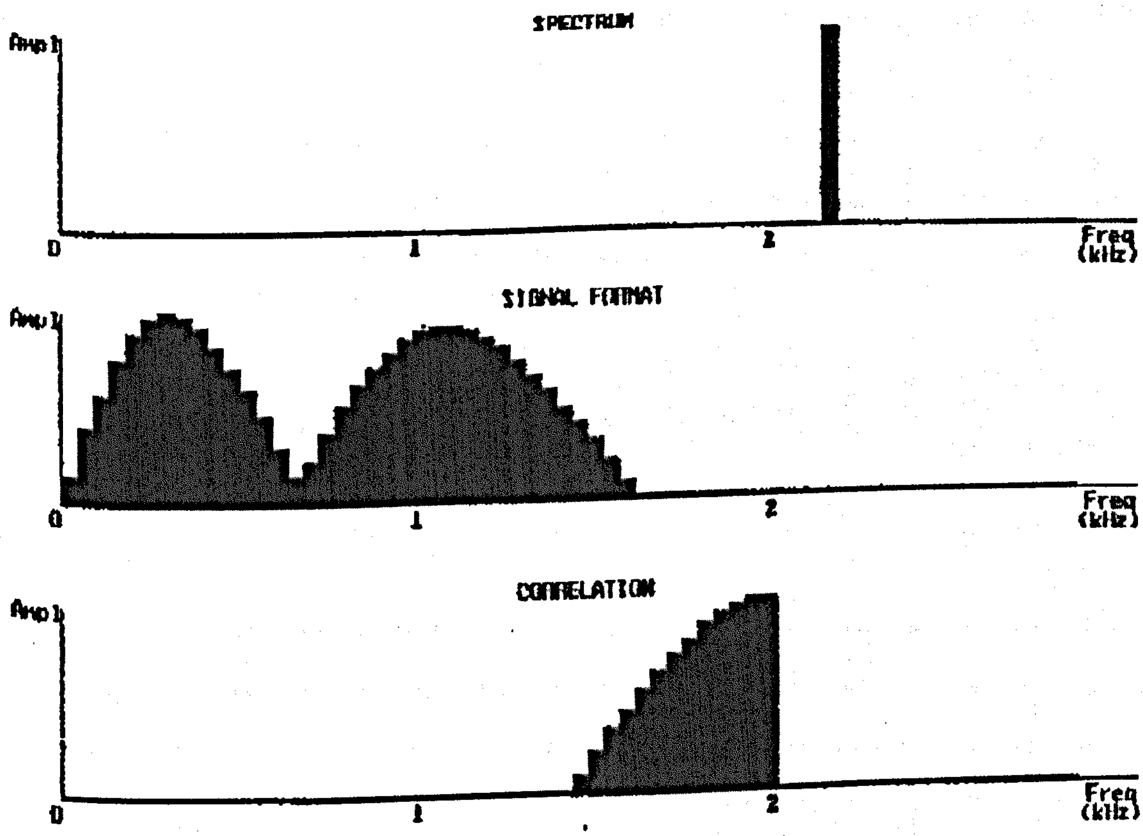


FIGURE 8.12b. TEMPLATE CORRELATION BENCH TRIALS RESULTS

FFT ANALYSER OUTPUT



SOFTWARE OUTPUT



(ii) Sinewave interference on higher modem tone.

Before template correlation was run:

Total number of blocks received	:	24
Number of blocks in error	:	22
Total number of bits received	:	1680
Number of bits in error	:	423
Bit error rate	:	25%

After template correlation was run:

Total number of blocks received	:	156
Number of blocks in error	:	2
Total number of bits received	:	9880
Number of bits in error	:	29
Bit error rate	:	0.3%

The output of the tempate correlation software and the FFT analyser was as shown by Figure 8.12b.

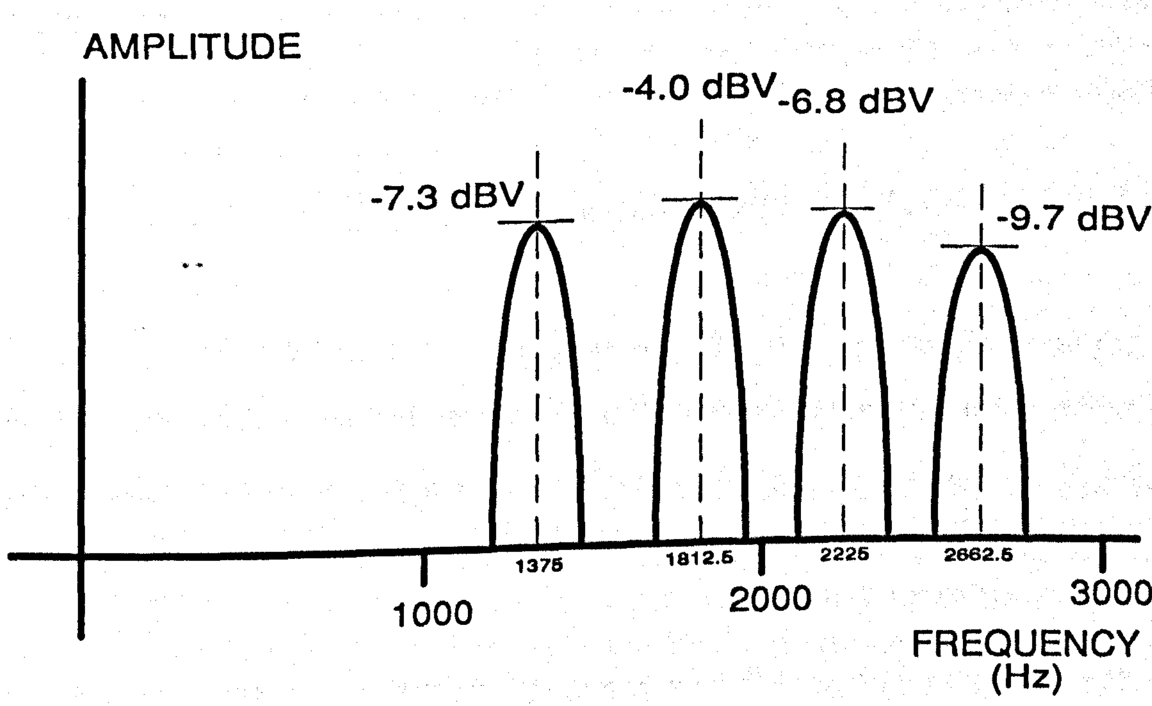
(iii) FSK modem interference.

The results before and after template correlation was run were as follows:

Total number of blocks received	:	0
Number of blocks in error	:	0
Total number of bits received	:	0
Number of bits in error	:	0
Bit error rate	:	Undefined

FIGURE 8.13. TEMPLATE CORRELATION BENCH TRIALS RESULTS

FFT ANALYSER OUTPUT



The output of the FFT analyser was as shown by Figure 8.13.

The above results are typical examples of those obtained in the full bench trials of the technique. In all but two of ten trials carried out with the sinewave interferer on the high modem tone the BER after template correlation was run was less than 1%, the BER before being usually between 20% and 30%. With the sinewave interferer on the low modem tone then little improvement in BER occured. The resulting BER improvement was in the range 1-4%, ie from say 25% to 21% at best.

8.5.4. Conclusions

Template correlation is able to reduce the bit-error rate of a communications channel due to interference, under certain conditions. The greatest performance enhancement occured with sinewave interferers being present initially on the high modem tone. In this case the template correlation software was capable of moving the interfering tone out of the receiver bandwidth, subjecting it to severe attenuation. This resulted in a significant BER performance improvement. In the case where the interfering tone was present initially on the low modem tone then the software was only able to move the tone (via de-tuning of transmitter and receiver) to a position in-between the two demodulator filter bandwidths. This inability to isolate the interferer completely was due to the requirement to keep the data transmission within the nominal 3 kHz bandwidth (in a practical communications system, this would be necessary in order to satisfy CCIR regulations and to avoid interfering with the adjacent channel).

It is felt that a greater performance enhancement would be possible if the modems used were more adaptive and that the demodulator filters had higher roll-off rates at the band edges. Further trials of the technique would thus require the construction of such a modem. As the time remaining on the research programme was insufficient to design an improved, flexible modem, trials of template correlation were terminated at this point.

In particular, the trials showed that the demodulator filters could not sufficiently attenuate in-channel interfering signals that were present outside the filter pass-bands. Also, the ability to move the modem tones independently of each other would have enhanced the performance of the system. In order to further evaluate template correlation it is recommended that an all-digital modem be constructed in software. This should allow steep band-pass filter roll-offs and the required modem tone frequency agility to be achieved.

8.6 Future Work

8.6.1 Trials

The initial bench trials described above showed that template correlation was, under certain conditions, able to improve the performance of a communications system operating over a channel with an interfering signal being present. However, it was felt that the modems used were not flexible enough to enable the full power of template correlation to be realised. Hence further trials of the technique are deemed necessary with more adaptive modems.

Ideally, an FSK modem, with the ability to move its tones independently of one another within the channel bandwidth, would be constructed. Also, the demodulator filters would have very high order roll-off rates at the edges of the filter pass-bands, in order to provide greater out-of-tone-bandwidth interference rejection.

In order to achieve a high roll-off rate and in-band frequency agility under computer control, it is thought that a software-implemented modem, employing digital filtering techniques in the demodulator would be optimum. A TMS-based implementation would facilitate easy interfacing with the template correlation software.

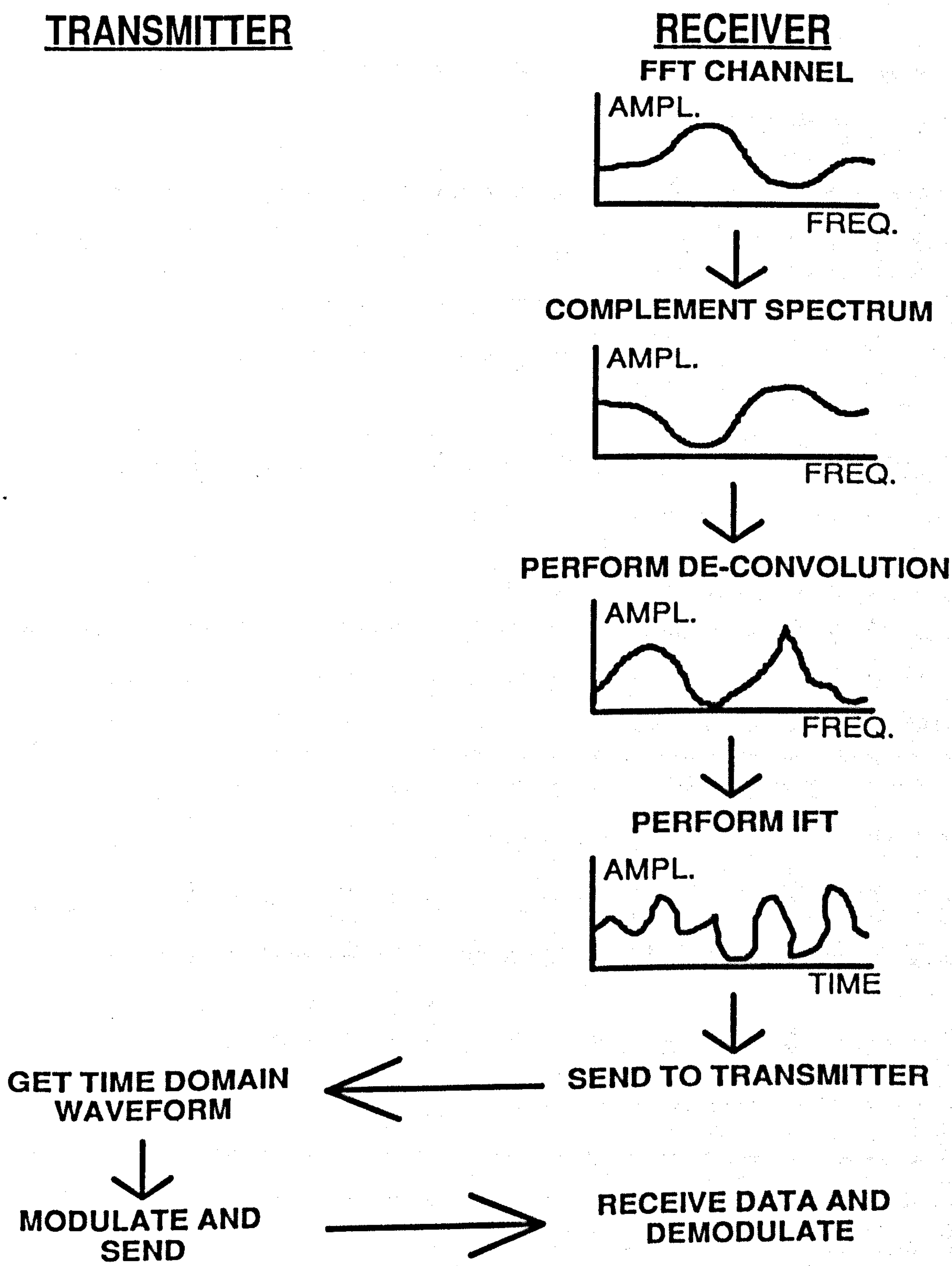
Once such a modem had been constructed and tested then further trials could be carried out. The ability of template correlation to reduce the bit-error rate of a channel under differing interference conditions could be assessed by recording several types of interference off-air onto tape (eg analogue speech, FSK, and facsimile transmissions) and then performing trials, using the apparatus and trials method described in Section 8.5.2. The effects of varying the transmission block length (this is, essentially, the same as varying the time between modulator tone position updates from the template correlation software) could be investigated in order to find the optimum modulation format update techniques for different interference conditions.

8.6.2 Spectrum Complementing

The advent of cheap, powerful microprocessors has been the enabling factor for many of the techniques described in this thesis. Processor technology is continuing to advance, with a large amount of research and development effort being spent on areas such as multi-processing, pipelined architectures and reduced instruction set computers (RISC's). Thus, in the future, it will be possible to embed a larger amount of processing power within each terminal of an HF communications system than is currently feasible.

Template correlation attempts to increase the overall throughput rate of communication systems operating over noisy channels via an approximation to the

FIGURE 8.14. SPECTRUM COMPLEMENTING ALGORITHM



maximum information rate condition of equation 8.17. Advances in processor technology will allow a closer fit to the ideal case to be made, in real time. A technique which adapts the signal format of a transmission to match the prevailing noise/interference conditions more precisely than template correlation, termed spectrum complementing, has been conceived, the details of which are presented below.

The algorithm of spectrum complementing is shown in Figure 8.14. Linear SSB transmission is assumed for the following description. The receiver obtains the noise/interference spectrum from the channel by performing an FFT on it. The resulting spectrum is then complemented to produce the spectrum of the ideal signal for that channel, as per equation 8.17. This spectrum is then subjected to deconvolution (a process which is described mathematically below) with the spectrum of the information to be transmitted (this would be a sinc(x)-type function for digital modulation schemes). An inverse Fourier transform (IFT) is performed on the resulting frequency domain representation to produce time domain samples which represent the "ideal" un-modulated signal. These samples are then sent, over engineering order wires (or the radio equivalent of them) to the transmitter which modulates the communications traffic on to this signal and then transmits it to the receiver. After modulation at the transmitter the frequency domain representation of this waveform will be the complement of the result of the FFT carried out initially by the receiver. Hence a closer approximation to the ideal maximum information rate condition will have been achieved than is possible with template correlation in the form described previously.

The process of deconvolution is achieved using matrices as shown below (Rhoads and Elastrom, 1968):

Modulation can be described in terms of a frequency domain convolution:

$$r(f) = \int_0^f m(f-u).x(u).du \tag{8.20}$$

where $r(f)$: the required signal after modulation.

$x(f)$: the signal before modulation (to be found).

$m(f)$: the modulating signal.

If $r(f)$ is finite, as in most practical examples, then an approximation to the above can be made. The integral of equation 8.20 can be broken down into a sum of integrals over N segments, each f Hz in length. Assuming constant excitation over each segment, then a finite sum can approximate to 8.20 as follows:

$$r(f) \approx \int_0^F m(f-u).x(u).du + \int_F^{2F} m(f-u).x(u).du$$

$$\ldots\ldots + \int_{(N-1)F}^{NF} m(f-u).x(u).du \tag{8.21}$$

For $0 < f < NF$

Thus:

$$r(f) \approx x(F)\int_0^F m(f-u).du + x(2F)\int_F^{2F} m(f-u).du$$

$$\ldots\ldots + x(NF)\int_{(N-1)F}^{NF} m(f-u).du \tag{8.22}$$

As $x(u)$ is constant over each segment and the u term increases as we sum across the waveform in increasing frequency, then equation 8.22 becomes:

$$r(f) \approx \sum_{n=1}^{N} x(NF).\int_{(n-1)F}^{nF} m(f-u).\ du \tag{8.23}$$

Evaluating $r(f)$ at each of the sampling points, $f = kF$, where $k = 1,2 \ldots N$.

$$r(kF) \approx \sum_{n=1}^{N} x(NF) . \int_{(n-1)F}^{nF} m(kF-u) . \, du \tag{8.24}$$

Now re-write the integral part of equation 8.24 and substitute λ for kF−u (change also the integral variable):

$$\int_{(n-1)F}^{nF} m(kF-u).du = \int_{(k-n)F}^{(k-n+1)} m(\lambda).d\lambda$$

$$= M(k-n+1) \tag{8.25}$$

ie the area under m(f) between (k−n+1)F and (k−n)F.

In order to satisfy causality, equation 8.25 will be zero if $k<n$.
Substituting equation 8.25 into equation 8.24 gives:

$$r(kF) \approx \sum_{n=1}^{k} x(NF) . \int_{(k-n)F}^{(k-n+1)F} m(\lambda) . \, d\lambda \tag{8.26}$$

This summation can be expressed in matrix notation as follows:

$$r = M.x \tag{8.27}$$

where r and x are the wanted modulated and un-modulated spectrum column vectors respectively, and M is the convolution matrix for the particular modulation scheme employed by the system.

The elements of the column vectors r and x are sequential samples of the frequency domain representations of the modulated and un-modulated signals respectively. The elements of the convolution matrix are the strip areas of the function m(f), ie the strip areas of the spectrum of the modulating signal.

Denoting column vectors as COL [ ] and letting F = 1 for simplicity's sake:

$$x = \mathrm{COL}\ [\ x(1)\ x(2)\ \ldots .\ x(N)\ ] \tag{8.28}$$

$$r = \mathrm{COL}\ [\ r(1)\ r(2)\ \ldots .\ r(N)\ ] \tag{8.29}$$

$$M = \begin{bmatrix} M(1) & 0 & 0 & .. & 0 \\ M(2) & M(1) & : & : & : \\ : & M(2) & M(1) & : & : \\ M(N-1) & .. & M(2) & M(1) & : \\ M(N) & .. & .. & .. & M(1) \end{bmatrix} \tag{7.30}$$

Element M(1) is taken to be the first non-zero strip area of m(f) – this corresponds to a frequency shift in the modulating signal.

The process of de-convolution, ie that which enables the required un-modulated signal to be determined from the ideal modulated signal spectrum and the modulating signal spectrum, can be achieved by inverting the matrix in equation 7.30.

Thus:

$$x = M^{-1}.r \tag{7.31}$$

The frequency domain samples of the un-modulated signal are thus contained in the column vector x.

This matrix-based technique is valid if there are no discontinuities in the modulated signal r, ie it it relatively noise free. Averaging r or filtering it should ensure that this criterion is met.

The main difficulty with the use of the above technique is the choice of modulation scheme. Phase-shift keying (PSK) of the time domain waveform would be easy to implement as both ends of the communications link have a replica of the un-modulated signal which could be used to achieve coherent demodulation. However PSK is a non-optimum choice for HF communications as HF channels are known to possess inherent phase instability. Amplitude-shift keying (ASK) could be used or, if more than one channel was available, FSK could be implemented with each channel effectively implementing one tone.

The computational requirements for spectrum complementing would be large, but it is expected that advances in digital signal processor technology (such as the forthcoming Texas Instruments TMS 320C30) would enable the technique to be implemented.

Additionally, the above technique could be used to equalise tones on a multi-tone modem. The amplitude of each tone would be set by the complement of the interference spectrum.

9 A NEW AUTOMATIC HF COMMUNICATIONS SYSTEM ARCHITECTURE

9.1 Introduction

Following the detailed description of two new FMS-enhancement techniques contained in Chapters 7 and 8, attention will now be turned to a description of the research carried out at an overall systems architecture level.

The current status of some automatic HF communication systems was reviewed in Chapter 5. The systems examined have a degree of commonality in that they use the same types of frequency management tools, eg embedded prediction models, RTCE etc., but their architectures and operational details are different.

The Plessey HF system evolved in response to funding from several different sources. In its initial form it consisted of a manually controlled system incorporating an embedded propagation model. Additional components were added as necessary and thus there was little structured design methodology employed during its construction.

Such organic growth tends to produce a system architecture consisting of individual software and hardware components linked together via complex and convoluted interconnections. If system growth continues in this fashion then a point is reached where the addition or subtraction of any constituent part of the overall system results in it behaving in a "problematic and quirky" manner (Sommerville, 1985). New ideas and techniques cannot be tested easily and it is thus difficult to adapt the performance of the system to changes in its operating environment.

It can be advantageous if an HF system is compatible with other communications media so that, during times of black-out and equipment failure, traffic can still reach its destination using other long-distance communications media, eg land-lines and satellite communications. This ability to successfully connect different hardware and software modules together would be particularly useful for communications facility managers who do in practice have several different communications media at their disposal.

Transparent, "telephone-style" operation is an ideal requirement: the user should not be aware of the underlying bearer system and its operation, as far as is practically possible. In this case, the user would enter the destination and other requirements for the message, such as its security requirements and latest tolerable arrival time, and the message itself. The system would then be left to cope with the complexities and difficulties of the HF medium, in order to ensure the reception and successful demodulation of the signal at its destination.

The aim of the work described in this chapter was to develop a structured HF system design methodology, in order to overcome the difficulties experienced by the author in adapting the Plessey system to incorporate the amateur-grade equipment in use at the Hull site and to try to achieve compatibility between individual HF systems and other long-range communications media. In pursuing the above goals, the operation of an automatic HF communications system was divided into two main tasks:

(i) FREQUENCY MANAGEMENT.

The aim of the frequency management routines is to ensure that the system is always operating using the optimum channel and equipment settings, eg bit rate, in-band tone positioning etc.

(ii) TRAFFIC MANAGEMENT.

Traffic management routines assume that the overall system is set up to operate in an optimum fashion. If the traffic is digitally encoded then the traffic management routines provide encryption/security services, character set translations, error-control management etc. When the traffic is analogue speech then traffic management is relatively easy. It may involve scrambling and, in a more advanced system, speech analysis and generation to provide language translation services.

A a co-ordinated HF systems' design methodology is presented in this chapter. This has been developed in order to meet the system requirements detailed earlier in this section.

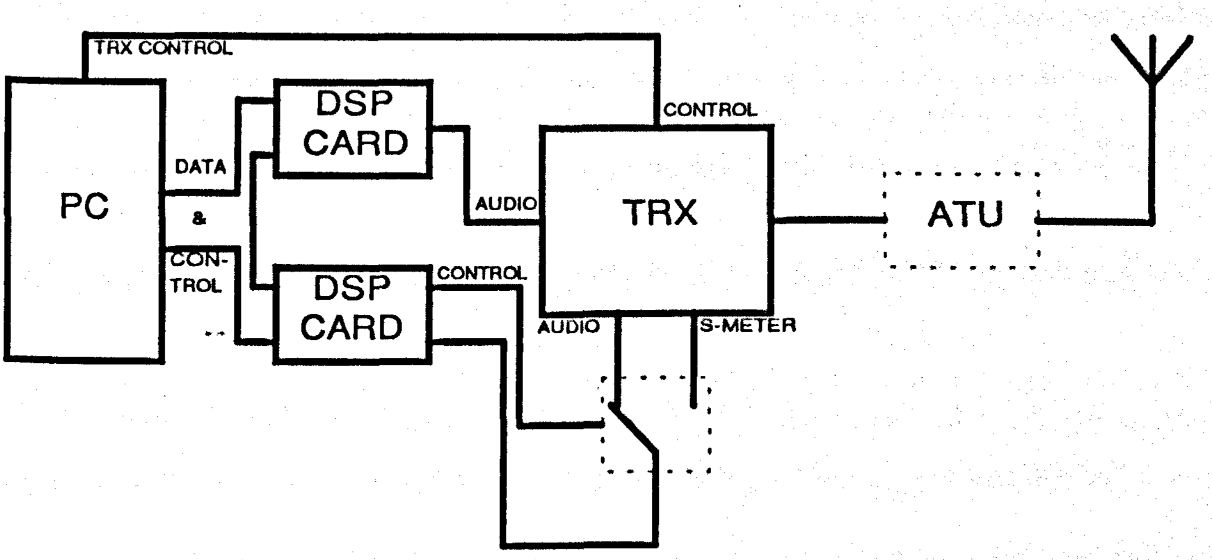
9.2 Hardware Configuration

Figure 9.1 shows the suggested hardware configuration for automatic HF communication systems' terminals. A description of the individual hardware units now follows.

The main, general purpose processing power in the terminal is provided by an IBM PC clone or a microcomputer system with similar power and abilities. An INTEL 80286-based machine (or better) with maths co-processor would provide the necessary processing power for quick and efficient execution of algorithmically intensive programs, such as PAPs.

Twin digital signal processing cards are included in the terminal configuration. These would most likely be based upon the TMS 320C25 processor and, for a pc-based implementation, they would be resident in the main processor's expansion slots. One card could be configured as a software modem thus enabling easy

FIGURE 9.1. TERMINAL HARDWARE



adaptation of data rate, in-band tone positioning, and the modulation format. Digital signal processors allow easy implementation of high roll-off rate digital filters. Such filters could be used to construct a demodulator with good out-of-tone-bandwidth interference rejection capabilities.

The remaining TMS 320 card would be used for miscellaneous signal processing activities such as, template correlation, chirpsounder and broadcast transmission monitoring, signal strength logging and interference analysis.

A computer-controllable transceiver, such as the transceiver used in the Hull terminal of the Plessey HF system (an ICOM IC 735), would allow fast and automatic tuning to be achieved. Due to the low cost of such amateur-grade equipment, it may be economically feasible to incorporate a separate receiver or transceiver into the terminal for signal monitoring and RTCE purposes. An automatic tuning unit would be included if a broadband antenna was not available at a terminal.

9.3 Frequency Management Software

9.3.1 Aims

The main aim of an HF FMS is to ensure that the overall communications system is using the optimum channel at all times. It is also responsible for adjusting equipment settings to avoid any in-band interference, thus minimising the BER, and also for choosing the optimum modulation format for a particular channel. A prediction of the likely time-variability of the chosen channel's interference profile is required to enable the timing of signal positioning adjustments to be determined.

In order to achieve the above aims the propagation, natural noise and interference conditions on all of the system's frequency allocations need to be known. The frequency management tools available to provide this information for an automatic HF communications system are:

(i) PAPs (see Chapter 3).

(ii) RTCE (see Chapter 4).

(iii) Passive monitoring (see Chapter 7).

(iv) Template correlation (see Chapter 8).

Ideally, the FMS should take into account the user's communication requirements, eg the amount of data, the destination, and the security level. Once these have been ascertained it should use a combination of the above tools to find the optimum channel and equipment settings to match these requirements.

An HF FMS has been generated which makes use of system design and programming techniques derived from the field of artificial intelligence. The resulting system is highly structured and it provides operational flexibility in that the one part of system's configuration can be adjusted without inadvertently affecting other elements. The FMS has been designed using a specialised type of expert system, known as a "multi-panelled blackboard system".

9.3.2 Expert Systems

A definition of an expert system is given by (Sell, 1985):

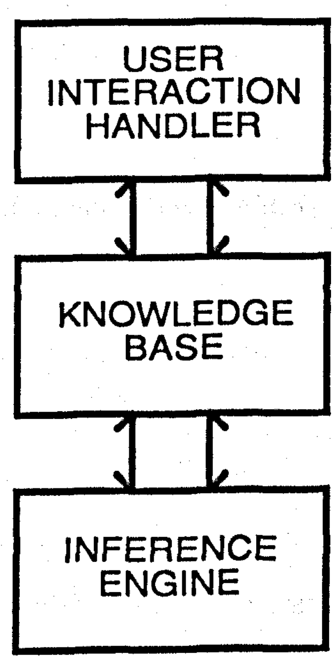
"An expert system is a knowledge-based system that emulates expert thought to solve significant problems in a particular domain."

They are computer programs embodying the expertise of one or more people in a limited subject area or domain with the ability to make inferences about that domain. They store the knowledge about a domain which has been extracted from a domain expert. Successful expert systems have been produced for a wide range of applications including the diagnosis of bacterial infections (MYCIN (Shortcliffe, 1976)), the evalution of geological prospects (PROSPECTOR (Duda et al, 1978)), and the configuration of VAX computers to meet specific customer's requirements (R1 (McDermott, 1980)). They differ from conventional software systems in that the data about a subject (termed the "domain specific knowledge") and the algorithms which use this data to produce the system's results (the inference mechanism) are kept entirely separate. Thus changes to the overall system's operation can be made easily, by relatively unskilled staff, without the need for re-compilation of the software.

The three main components of an expert system are shown in Figure 9.2 (Turing Institute, 1987). The user interaction handler is responsible for the acquisition of the knowledge from a domain expert: in the case of the MYCIN expert system above, the domain expert would be a medical doctor. The user interaction handler also allows the user of the system to ask "how" or "why" a decision was made. This information is invaluable to the designer when debugging an expert system.

The knowledge base stores the domain specific knowledge in a form that is readable and recognisable to the domain expert, in order to further enhance the ease of de-bugging of the system. Knowledge is usually represented in one of the forms detailed below:

FIGURE 9.2. COMPONENTS OF AN EXPERT SYSTEM



(i) Complex record-like structures (as in the PASCAL programming language) allowing the expression of the inter-dependency of data items.

(ii) Rules, ie simple IF-THEN-ELSE type statements.

(iii) Fuzzy logic, ie facts can be "probably" or "possibly" true, as well as just simply true or false.

(iv) Uncertainty, ie the expression of facts or domain knowledge with a confidence level associated with them in the form of a probability figure for the fact being true.

The inference engine works on the knowledge base as would an interpreter on a programming language. It produces code which allows the expert system to run on the underlying hardware.

The two main solution search strategies used by expert systems are "forward chaining" and "backward chaining" (Turing Institute, 1987). A forward chaining expert system uses the facts that are known about a particular domain at a given time, together with the relevant rules, to work towards the solution. A backward chaining expert system starts with a solution to the problem and then tries to justify this using the rules and the known domain data. Which methodology is used by a specific expert system depends largely upon the nature of the problem to be solved.

9.3.3 Blackboard Systems

Blackboard systems are a specialised type of expert system. Generally, they are used to solve programming problems possessing the following characteristics (Nii, 1986):

(i) Complexity.

(ii) The need to integrate diverse data types.

(iii) Unreliable and uncertain data.

(iv) The need for an evolutionary solution.

(v) The need for multiple lines of reasoning.

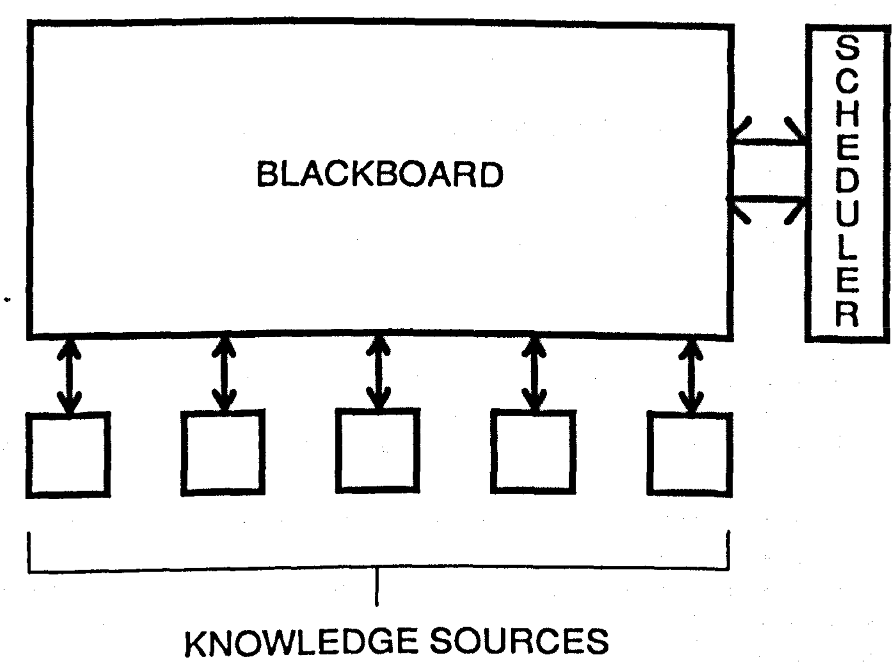
(vi) The need for several independent sources of information to co-operate in producing a solution.

(vii) The need for opportunism, ie a combination of forward and backward chaining techniques, in the problem solving methodology being applied.

Figure 9.3 shows a generalised model of a blackboard system based on the Hearsay II Speech Understanding System (Nii, 1986). There are three main components: the knowledge sources, the blackboard and the scheduler. The operation of the system is explained below.

The modus operandi is analogous to a group of people trying jointly to solve a jigsaw puzzle. The blackboard is the surface on which they are placing pieces of the jigsaw that fit together with the other pieces already there. Each person solving the puzzle is represented by a knowledge source for the purposes of this description. All of the puzzle solvers look at the completed "islands" of puzzle and the remaining, un-attached pieces in order to try and fit them on to the board. When a piece is found that fits, the solver then asks the scheduler to allow him to

FIGURE 9.3. COMPONENTS OF A BLACKBOARD SYSTEM



place the piece on the blackboard (only one piece may be added at once). The only communication between individual solvers is via the blackboard, ie in this example by placing pieces of jigsaw on the board.

The underlying hardware on which a blackboard system is implemented influences the system's configuration. In a multi-processor computing environment each knowledge source could be allocated a processor of its own: there would thus be no need for the scheduler as each knowledge source would add to the board when it had some information that would lead towards the solution of the problem being attempted.

Blackboards are particularly well suited to solving problems that are ill-defined. They allow the postponement of design decisions and they enable the system configuration to be adjusted easily.

9.3.4 A Blackboard-Based HF FMS

Figure 9.4 shows an HF FMS design implemented on a multi-panelled blackboard model. There are four blackboard panels in total: the main panel, the PAP panel, the RTCE panel and the template correlation (TC) panel. The main panel is fed information from the other three blackboards and it is responsible for invoking each of the panels, as required. A description of the architecture and operation of the FMS design now follows.

(i) MAIN PANEL

Figure 9.4a shows the main panel. It is responsible for providing a performance rank of the system's frequency allocations in response to the user's communications objectives. This rank would then be used to

FIGURE 9.4

A BLACKBOARD-BASED HF FMS

NB. INPUTS TO KNOWLEDGE SOURCES ENTER FROM THE BOTTOM OF THE PAGE-
OUTPUTS FROM THE KNOWLEDGE SOURCES LEAVE FROM THE TOP

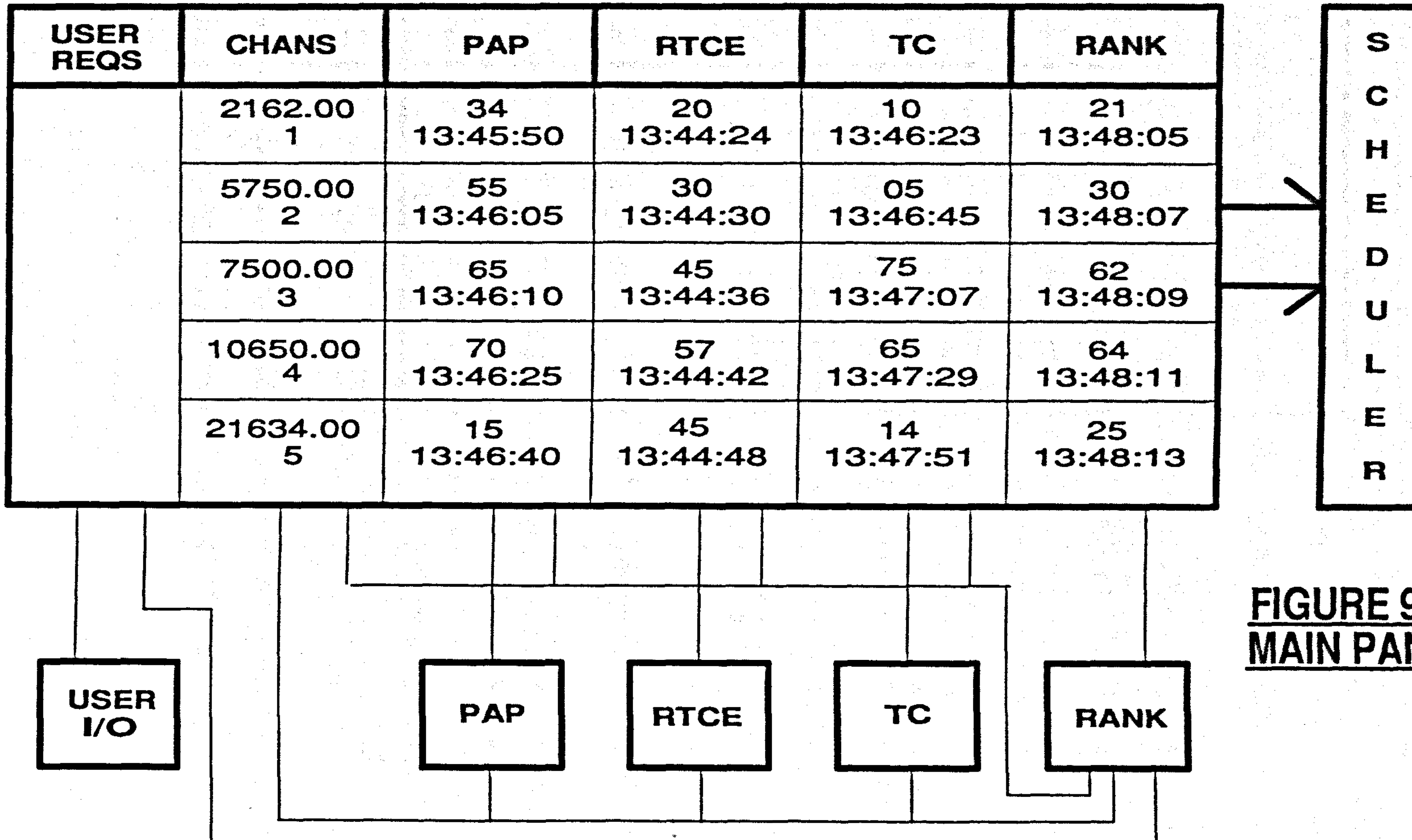


FIGURE 9.4a
MAIN PANEL

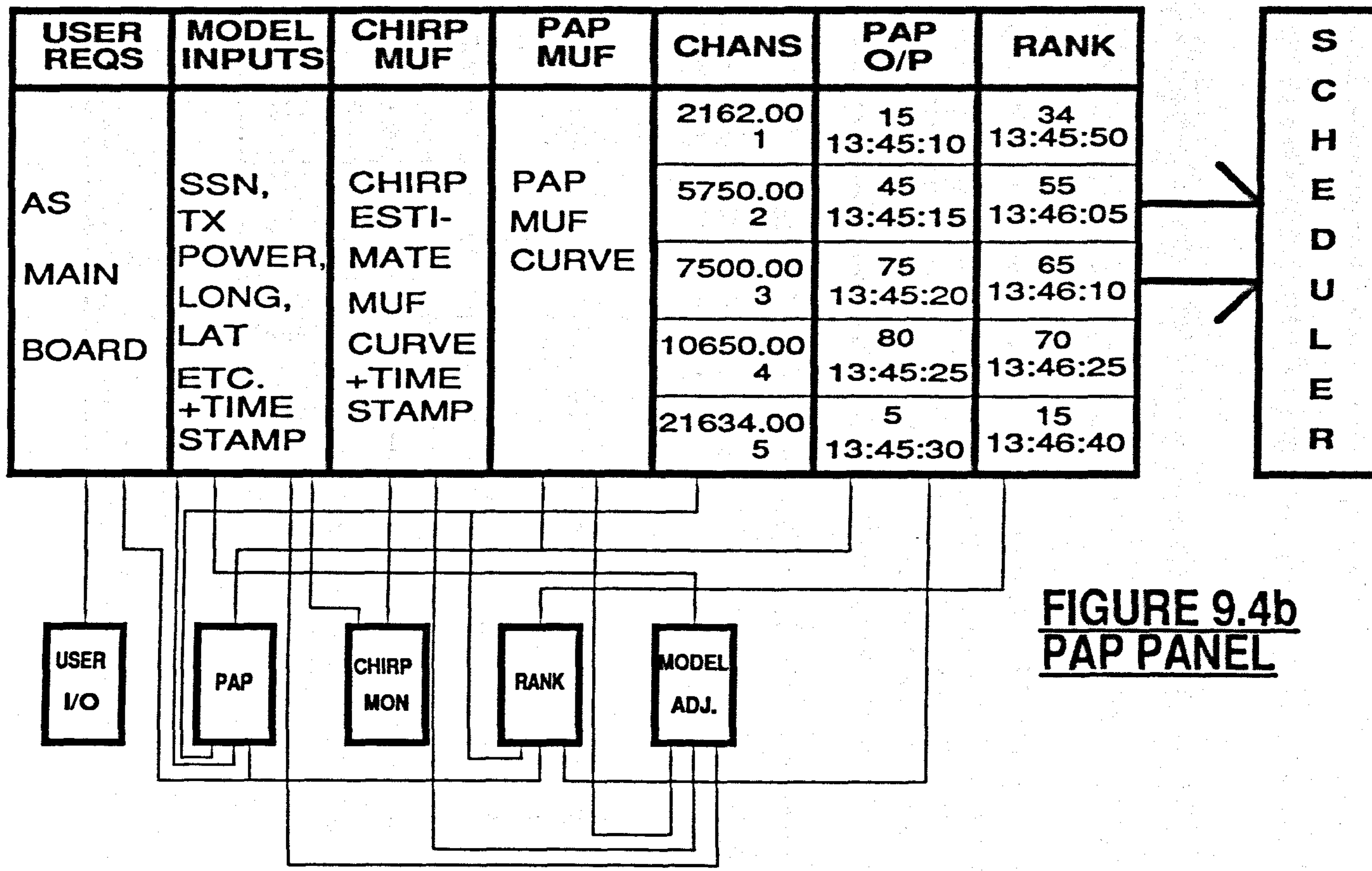


FIGURE 9.4b
PAP PANEL

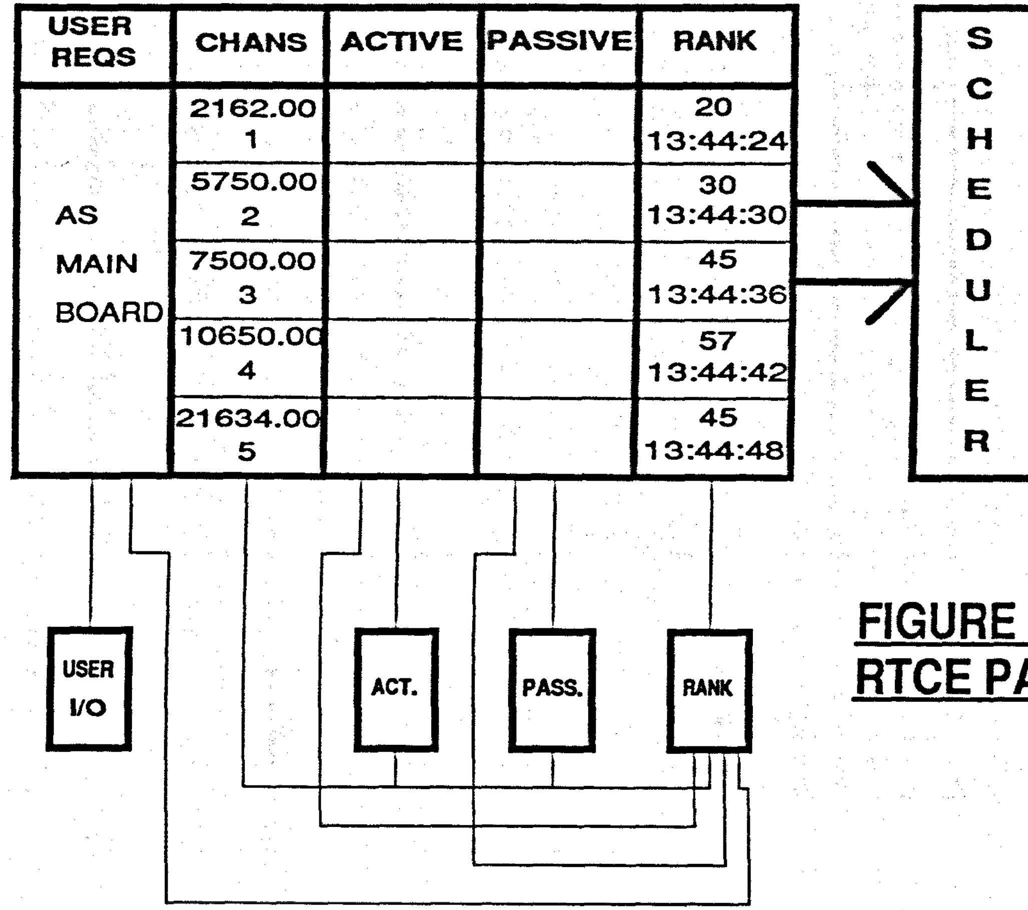


FIGURE 9.4c
RTCE PANEL

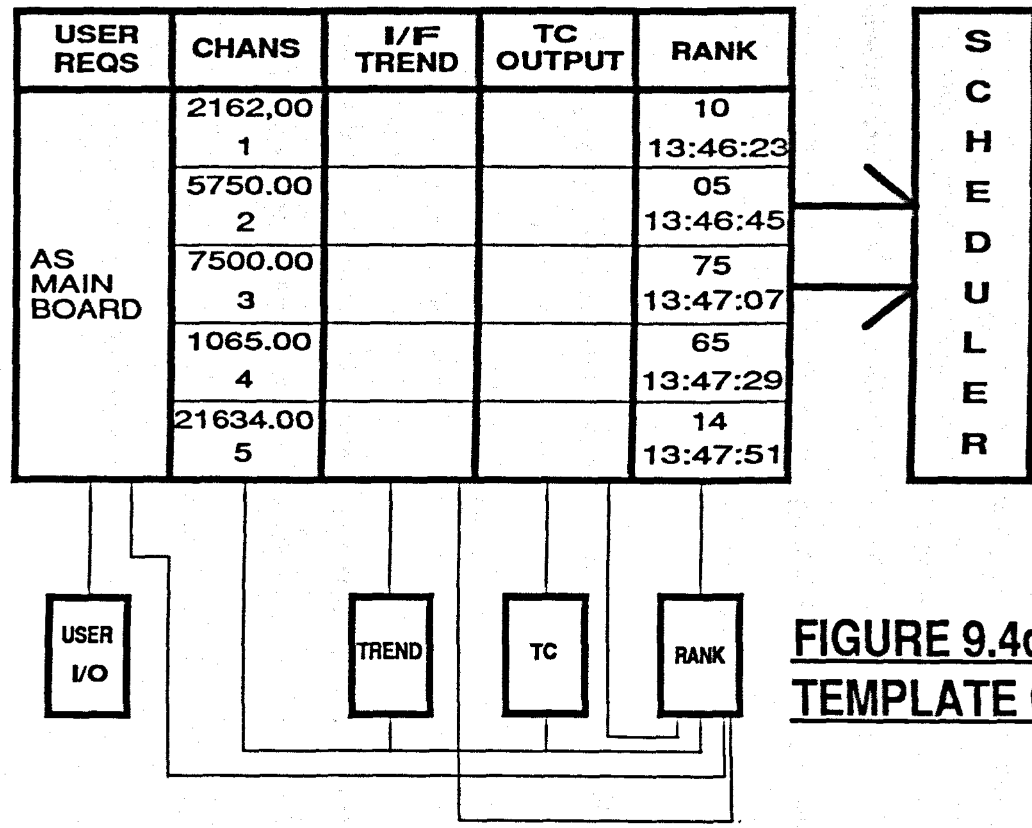


FIGURE 9.4d
TEMPLATE CORRELATION PANEL

select the system's operating frequency.

BLACKBOARD

The blackboard itself is split into six fields. The "USER REQS" field holds information regarding the user's communication requirements, eg the amount of data, the required security level for the message, the destination of the message etc. This information is accessed by the knowledge source "USER I/O.

The field "CHANS" contains the user's frequency allocations. The first entry is the frequency of the allocation in kHz, the second being an arbitrary channel number.

The "PAP", "RTCE", and "TC" fields contain performance parameters for each of the frequency allocations resulting from the execution of the knowledge sources "PAP", "RTCE" and "TC". The data present on the blackboard for each channel consists of a two digit number, representing a percentage rank figure given to a channel by the knowledge source, and a "time-stamp", logging the time at which the rank figure was written to the board. This time-stamp is used during the scheduling of the knowledge sources. The higher the rank percentage, the higher the performance rating of a channel by a knowledge source.

The field "RANK" contains the combined rank percentage for each channel, computed from the three rank scores for PAP, RTCE and TC. The data entry format is the same as for the other ranked fields. The combined rank is computed by the knowledge source "RANK". In Figure 9.4a the combined rank is an average of the PAP, RTCE and

TC ranks. In a practical implementation of this system other, weighted combinations of the ranks could also be employed.

KNOWLEDGE SOURCES

"USER I/O" is concerned with interaction between the user and the system. It extracts information such as the type of service required, the destination for the message and any special routing requirements.

"PAP", "RTCE", and "TC" are all panels of the overall FMS blackboard system. They produces rank figures for the system's frequency allocations as their output.

"RANK" combines the individual rank percentages for each channel to produce an overall rank figure. The optimum channel, ie the one that is most likely satisfy the user's communication requirements, is the one having the highest rank.

SCHEDULING

The schedule for the main board would depend upon the operational requirements of the user. In the period before communications traffic is passed PAP and TC would be scheduled concurrently (TC runs at the receiving station, whilst PAP can be run at either end of the path). The scheduler would also be responsible for making sure that the data on the blackboard was up-to-date before any channel selection decisions were made.

(ii) PAP PANEL

Figure 9.4b shows the PAP panel of the FMS blackboard. The aim of this sub-system of the FMS blackboard is to supply the main panel with the channels' predicted performance in the form of percentage figures, according to the likely propagation conditions on each one.

<u>BLACKBOARD</u>

The blackboard is divided into seven fields, two of which ("USER REQS", and "CHANS") are identical those on the main panel. The field "RANK" corresponds to the field "PAP" on the main panel.

"MODEL INPUTS" contains the input parameters for the knowledge source "PAP". The types of data that would be resident in this field are covered in Chapter 3 of this thesis. Also, the details of the chirpsounder transmitter used for chirp-based MUF estimations are stored here. This data enables model accuracy enhancements to be made.

"CHIRP MUF" contains a 24 hour MUF profile for the control path from a chirpsounder transmitter used to enhance the accuracy of the PAP. A time-stamp is also written to this field. This information could be produced by the chirpsounder MUF estimation system described in Chapter 7 of this Thesis. It has been assumed, for the purposes of this design, that the PAP will have its accuracy enhanced by the sun-spot number manipulation method detailed in (Uffelman et al, 1984).

"PAP MUF" contains a 24 hour MUF profile, generated by the PAP, for the same path as the above "CHIRP MUF" field. Again a time-stamp is included in the field.

"PAP O/P" contains the reliability figures for each channel for the path in question, ie the percentage of days in the month that the required path will propagate at that frequency. A time-stamp is also included to aid scheduling. This information can be used directly as the PAP rank figure or, as in this example, it can be weighted in some way by the knowledge source "RANK" before being written as the rank.

KNOWLEDGE SOURCES

"USER I/O" is the same as that used on the main panel.

"PAP" is a propagation analysis procedure, of the type described in Chapter 3. It takes as its input the user's requirements, the model input parameters and the system's frequency allocations. It produces reliability figures for the path and frequencies as its output.

"CHIRP MON" represents the chirp-based MUF estimation system, detailed in Chapter 7. It produces a 24 hour MUF profile for the path from the selected chirpsounder source. Also, it writes the details of the chirpsounder transmitter to the "MODEL INPUTS" field so that the knowledge source "PAP" can be run on the same path for model accuracy enhancement purposes.

"RANK" takes as its inputs the PAP reliability figures, and the channel details and produces a rank for the channels in the same format as the field "PAP" on the main panel.

"MODEL ADJ" is responsible for enhancing the accuracy of the PAP. It achieves this using the methods detailed in (Uffelman et al, 1984), ie the PAP-generated MUF profile is adjusted by manipulating the PAP input parameters and repeatedly re-running the PAP so that a best-fit to the chirpsounder-generated MUF profile is achieved. The values of input parameters used to achieve this best fit are then used in subsequent propagation predictions until the accuracy enhancement procedure is carried out again.

SCHEDULING

The operation of the PAP panel is divided into two parts: model accuracy enhancement and channel propagation prediction.

During the model accuracy enhancement stage, the knowledge sources "CHIRP MON", "PAP", and "MODEL ADJ" will be activated. "MODEL ADJ" will wait until "CHIRP MON" and "PAP" have produced corresponding 24 hour MUF profiles and it will then be scheduled so that an attempt to enchance the accuracy of the propagation model can be made.

During the channel propagation prediction phase, knowledge sources "PAP" and "RANK" will be activated. Once "PAP" has finished executing then "RANK" will be scheduled.

(iii) RTCE PANEL

Figure 9.4c shows the details of the RTCE panel. The aim of this blackboard system is to provide the main panel with channel rank figures derived from RTCE output.

BLACKBOARD

The blackboard has been divided into five fields. The fields "USER REQS", and "CHANS", are identical to those on the main panel. Field "RANK" correponds to the field "RTCE" on the main panel.

"ACTIVE" contains information derived from any active RTCE that has been performed on the system's channels, eg a BER figure for the channel. A time-stamp is written to the field in order to aid scheduling.

"PASSIVE" is the field containing any information derived from passive RTCE routines, eg the packet repeat-request rate in an ARQ scheme.

KNOWLEDGE SOURCES

"USER I/O" is the same as its namesake on the main panel.

"ACT" contains the active RTCE routines embedded within the system.

"PASS" contains the passive RTCE routines embedded within the system. Examples of both active and passive RTCE techniques are given in

"RANK" takes as its input the user's requirements, the channel numbers and frequencies, and the outputs from passive and active RTCE. It uses this information to rank the channels, the channel which is most likely to match the user's requirements receiving the highest rank.

SCHEDULING

"ACT" and "PASS" are run according to the mode of operation of the terminal, ie if a call is in progress then "PASS" will be used ("ACT" could also be employed if duplicate RF equipment was available at the terminal); when the terminal is in a dormant state "ACT" will be scheduled. "RANK" is scheduled whenever a time-stamp in fields "ACTIVE" or "PASSIVE" is found to be later than the corresponding time-stamp in the field "RANK", thus making sure that the information in field "RANK" is always up-to-date.

(iv) TEMPLATE CORRELATION PANEL

Figure 9.4d shows the details of this blackboard panel. The aim of this blackboard sub-system is to try and match the user's transmission requirements, eg required bit rate, BER etc, by using template correlation to provide the decision data.

BLACKBOARD

The blackboard has five fields. The fields "USER REQS", and "CHANS" are identical to those on the main panel. Field "RANK" corresponds to "TC" on the main panel.
"TC OUTPUT" contains the results of performing template correlation on a channel. If the terminal has only one type of modulation available, then template correlation would be used to find the channel with the minimum potential BER. In this case the field would contain the estimated BERs for the channels. If a choice of modulation formats was available then this field would detail the modulation formats and their corresponding BERs for each channel.

"I/F TREND" contains information regarding the likely trends in the interference structure of each channel. This information would be used to specify the interval between signal format adaptation and modulation scheme changes so as to minimise the BER.

KNOWLEDGE SOURCES

"USER I/O" is the same as in the main panel.

"TC" contains the template correlation routines described in Chapter 8 .

"TREND" contains the routines necessary to analyse the interference structure of the channels with respect to frequency and time, eg FFT routines, averaging algorithms, signal identification schemes etc. It writes interference trend information to the blackboard.

"RANK" takes as its inputs the user's requirements, the channel frequencies and numbers, the interference trend information, and the output from template correlation. It compares the abilities of the channels with the user's requirements and produces a rank of the channels from this comparison.

SCHEDULING

Knowledge source "TREND" would be scheduled to run when the terminal was not transmitting or receiving communications traffic.

"TC" would run when the interference trend information, written in field "TREND", dictated that a signal format update was necessary. The scheduler would be responsible for reading field "TREND" and invoking knowledge source "TC" as required.

"RANK" would be run when any of the time-stamps in field rank were earlier than those in field "TC OUTPUT".

9.4 Traffic Management Software

9.4.1 Introduction

The traffic passed by HF communication systems is usually either analogue speech or digital, binary data. A traffic management regime for an HF system assumes that the FMS has found the optimum channel for the particular message requirements. However, because of the nature of HF paths (see Chapter 2) errors will be induced on the channel due to bursts of noise and interference and also due to propagation related effects, eg auroral black-outs. Analogue speech communications can cope with short burst errors relatively well. The humans at each end of the link possess the processing power and abilities to fill in gaps in the received messages due to such burst errors. However, burst errors can be catastrophic for digital communication systems. Thus there is a need to provide error-control facilities for HF traffic. However when the propagation path fails completely, then neither analogue speech or digital data can be received and demodulated successfully.

The traffic management scheme employed in an automatic HF system may be required to provide user-oriented services, eg for digital data, encryption, character set translations, network routing, and for scrambling of an analogue speech message to avoid eavesdropping.

Traffic management is relatively simple for analogue speech. At its most complex it involves the use of waveform companding, spectrum equalisation or scrambling techniques. Thus, this section of the thesis now concentrates on traffic management for digital data transmissions.

9.4.2 Traffic Management for Digital Data

Transparent, telephone-style operation is a requirement for automatic HF communication systems. The user should not be (unwillingly) made aware of the underlying system's operations. As stated in the introduction to this chapter compatibility with other long-range communications media would be advantageous, allowing the re-routing of communications traffic when required. Additionally, a structured design methodology is neceassary in order to facilitate easy modification of the system' configuration and thus to enable it to adapt to changes in its operating environment. Also, provision of user-oriented services and link flow control algorithms is desirable.

The International Standards Organisation Open Systems Interconnection (ISO-OSI) model for data transfer between computing equipment was designed with the above aims in mind. It was developed to provide a framework for the definition of standard protocols for information networks (Zimmerman, 1980). In the late 1970s individual computer manufacturers were developing their own interconnection and networking standards which were generally incompatible with each other. This meant that customers became "locked-in" to buying equipment from one manufacturer, a fact that was seen by the major computer suppliers as discouraging to potential customers. Global networking of computer systems would have been impossible in this scenario and thus the abilities of information retrieval systems, eg the INSPEC database, would be severely limited.

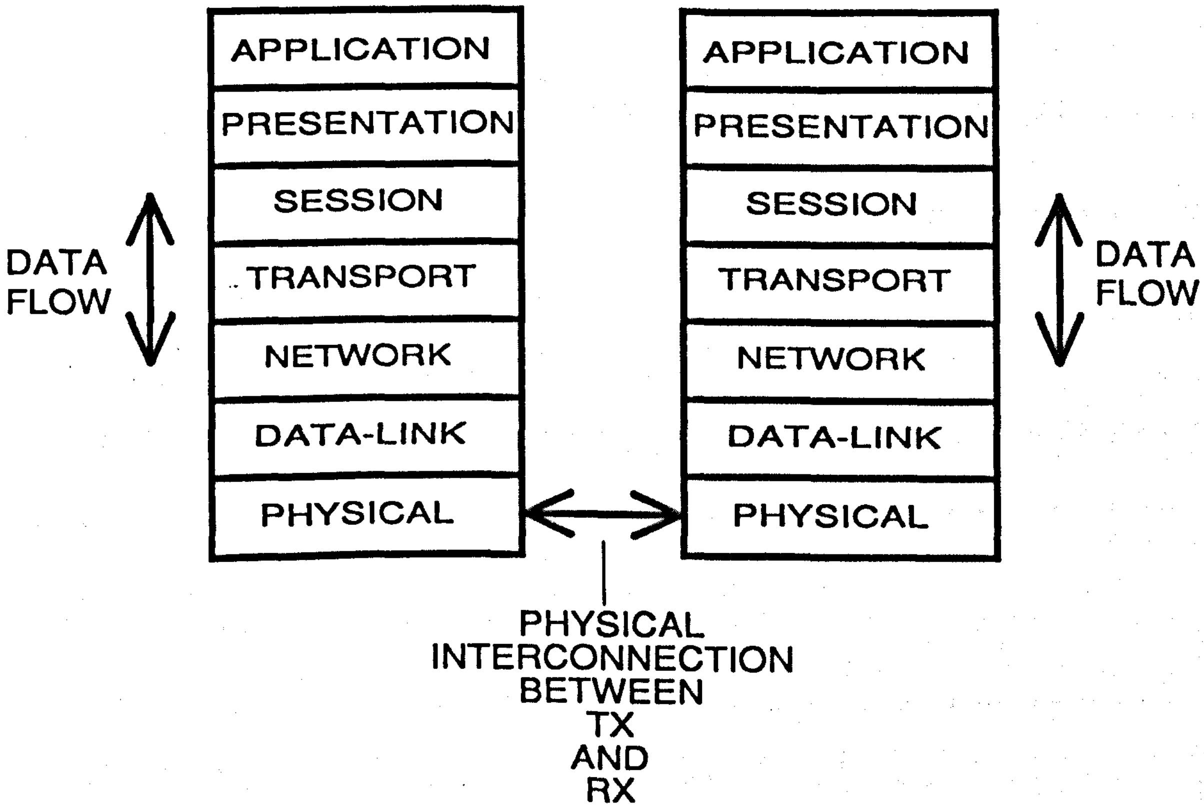
Figure 9.5 shows the ISO-OSI 7-layer model (Zimmerman 1980). Each layer should add to the services provided by the lower layers and a new layer is created when a new level of abstraction is needed. The interfaces between the layers are precisely defined to minimise the inter-layer data flow. A description of the layers now follows (Tanenbaum, 1984).

Referring to Figure 9.5:

PHYSICAL — This is responsible for the transmission of the raw data bits over the communications channel. It is concerned with the mechanical, electrical, and procedural interfacing to the network.

DATA-LINK — This layer provides an error-free channel for the Network Layer by adding error-control to the raw, bit-level service provided by the physical layer. It is also responsible for

FIGURE 9.5. ISO-OSI 7-LAYER MODEL



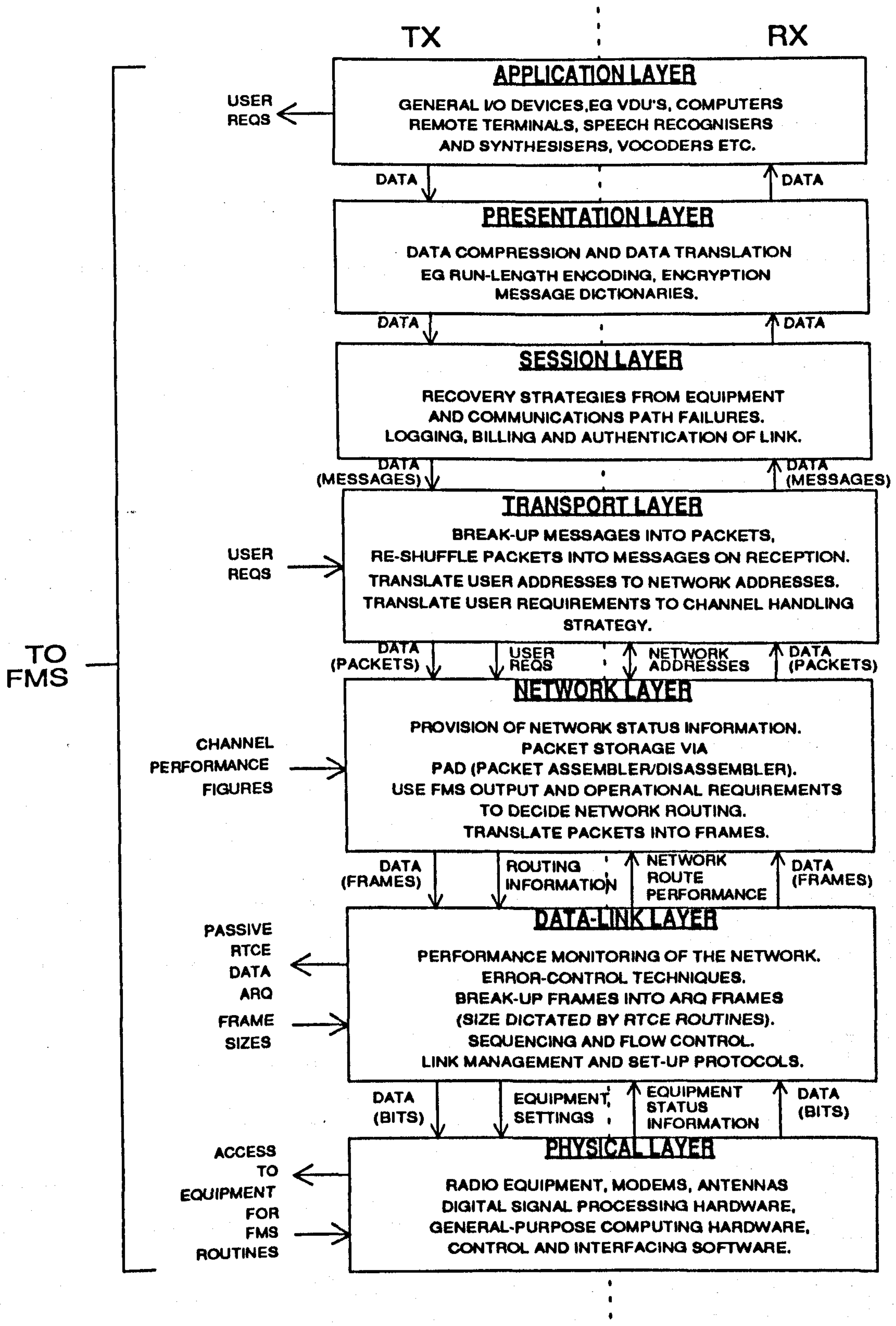
link set-up procedures, the provision of link management and link protocols.

NETWORK	The primary concern of this layer is the routing of the communications traffic through the network.
TRANSPORT	The translation of the user's communication requirements to a channel handling strategy, and the translation of user addresses to network addresses are the main tasks performed by this layer.
SESSION	This layer is concerned primarily with user authentication, call logging and the billing of the user.
PRESENTATION	Data translation and encryption facilities are offered by this layer.
APPLICATION	This layer is the user's own process that requires network communications.

Generally, the data is split up into progressively smaller pieces as it passes from the user's process in the Application Layer to the Physical Layer where it appears as individual bits.

The ISO-OSI model has been applied to HF communications in order to provide traffic management services. Figure 9.6 shows the resulting model. This is a revised version of that which appeared in (Jowett, 1987). The HF-specific features of the new model are discussed below. The transmission case is detailed only. Reception involves the reversal of the processes described below.

FIGURE 9.6. OSI 7-LAYER MODEL APPLIED TO AUTOMATIC HF RADIO COMMUNICATION SYSTEMS



The user enters his/her own specific communications requirements, eg data destination addresses, security and encryption requirements etc, via the Application Layer. These requirements are then passed on to the FMS to enable it to predict the likely optimum channel for the required link.

The FMS passes part of the user's requirements back to the traffic management system. The network addresses of the destination station(s) are required by the Transport Layer so that it can append destination information to the data packets. This enables traffic to be re-routed via intermediary stations when a direct path from source to destination is unavailable. The user's requirements are also used to select the optimum channel handling strategy for the particular path, ie the types of ARQ and error-control schemes to be utilised.

In the Network Layer, the channel rank from the FMS, the user's requirements from the Transport Layer, and the network performance information from previous communication sessions over the same path are used to determine exactly how the traffic should be routed through the newtork.

The Data Link layer extracts passive RTCE data, possibly in the form of an ARQ block repeat-request rate (see Chapter 4 for a discussion of this and other passive RTCE techniques). The RTCE routines monitor the quality of the channel currently in use and this information is used to determine the optimum size of data blocks for the link ARQ scheme (if used).

The Physical Layer provides access to the terminal hardware for all traffic and frequency management routines. Thus the FMS algorithms which measure the quality of the system's frequency allocations use it to control the terminal's equipment and to extract link performance data.

The proposed design thus consists of traffic and frequency management routines which operate in a co-ordinated and integral fashion.

9.5 Concluding Remarks

An HF communication system design methodology has been described which makes use of AI programming techniques in order to give operational adaptivity and ease of maintenance. The FMS design described is an improvement over current systems, having the benefit of passive, chirpsounder monitoring, template correlation and a flexible, but structured, architecture. The overall communications system would appear to be transparent to the user, this being achieved through the use of the OSI-ISO seven-layer model as a traffic management system framework .

A decrease in the price/performance ratio of microprocessor technology and the availability of cheap, adaptive radio equipment make such a design feasible. Implementation of this design would require the construction of a new system and radio trials.

In contrast, the next chapter presents a radically different approach to the design of automatic HF communication systems. The proposed terminal architecture incorporates an FMS which is based on a learn-by-induction expert system. Operational data, eg prediction model output, link performance parameters, model input figures etc., derived from trials of a successful HF communications system, is used to generate "rules" about the ideal behaviour of a generalised HF system. These are then manually edited by the system's designers and are used as the knowledge base for the expert system implementation.

The proposed design would have the ability to adapt its behaviour according to the specific characteristics of a particular HF path, via new rules generated as the system operates. This would enable HF users, communicating over paths exhibiting large variations in operational conditions, eg those operating from mobile platforms or from within polar regions, to optimise the performance of a link and thus to increase the overall traffic throughput.

10 AN ALTERNATIVE HF COMMUNICATIONS SYSTEMS ARCHITECTURE

10.1 Introduction

The HF communication system architecture described in Chapter 9 incorporated many of the components used by current automatic HF communication systems (see Chapter 5). The proposed architecture of the overall HF communications system should improve its maintainability and ease of adaptation. A degree of compatibility with other, long-range communications media may be achieved via use of the OSI seven-layer model.

Generally, the architecture described represents a re-structuring of current HF communication systems with some extra "tools", in the form of template correlation and chirpsounder monitoring routines, being added to help overcome the deficiencies of current systems. A radically new approach to the design of automatic HF communication systems is proposed in this chapter.

An architecture is described in this chapter which is radically different from that proposed in Chapter 9. It makes use of passive and active measurements of the conditions prevailing on the system's allocated channels. Use is made of the processing power embedded within the communications terminal and artificial intelligence programming techniques, in the form of a learn-by-induction expert system, to maximise the amount of information extracted from these measurements and try to enhance the performance of the overall system.

The proposed system learns from its operating environment and can adapt its behaviour to changes occuring in operating conditions. Such adaptibility will be of value to HF users operating with highly variable path conditions, eg users operating from mobile platforms and those operating in polar regions. It is also anticipated that this new approach to HF system design will enable rule-based models of an HF path to be produced, which can be used in other communications systems.

10.2 Machine Learning Techniques

10.2.1 Learning and Intelligence

Human knowledge and intelligence are difficult concepts to define (Hart, 1989). Human knowledge includes some basic rules, usually learnt from experience. Humans also have the "gifts" of insight and intuition. For example, when solving a problem of a type that has not been met before, they have the ability to cover missing patches of data using generalised knowledge about what has been the case in any similar types of problem that have been previously encountered. Also, "flashes of inspiration" have led to many of the world's greatest discoveries. Additionally, there is much evidence to support the theory that human knowledge is implicit in the structure of the neural networks within the brain (McClelland and Rumelhart, 1986).

Many researchers have attempted to decide whether or not a computing system can possess intelligence. Hypothetical examination methods, which are used as intelligence criteria, have been devised: one example of these methods is the Turing Test (Turing, 1950). However, these tend to be impractical and in-conclusive. It is unlikely that human-style intelligence will be achieved using currently available hardware and software, because of the following reasons.

(i) Computers lack a semantic understanding of the problems they are programmed to tackle. They are thus effectively valueless and motiveless when solving problems. Also, computers cannot, at present, model the intuition and inspiration that humans use to cover the gaps in their knowledge.

(ii) There are basic differences between the ways in which humans and computers tackle problems (Hart, 1989). Human reasoning involves what is known generally to be the case, ie general knowledge resulting from years of experience, intuition, and any domain-specific knowledge that is present, whereas, computer-based reasoning tends to use general techniques to detect patterns in the domain-specific data.

Hence, it is currently impractical to expect true, "human" intelligence from a machine. However, computers can act as "informed assistants" (an alternative definition of an expert system), extracting knowledge from human experts, analysing it and enabling it to be accessed quickly and reliably. The inherent objectivity of computers allows them to see patterns in data that humans miss: their lack of semantic understanding aids them in achieving this. However, because of this lack of understanding they can produce inappropriate results due to the fact that the data analysis is performed without the benefit of any general knowledge or intuition.

Extracting knowledge from experts and implanting this in a computer in a meaningful way (a process known as "knowledge elicitation") is notoriously difficult. Often, a domain expert will not be able to put his/her knowledge into words: the knowledge may be implicit and it may enable the expert to solve a problem successfully without the knowledge of why a particular solution method works: it was successful for previous examples of this type so it is used for future problems of a similar type. The difficulties associated with extracting expert knowledge are known collectively as the "knowledge elicitation bottleneck". One way to alleviate the problems associated with knowledge elicitation is to let the computer learn by induction or experience, as humans do. This involves, at the simplest level, the drawing of generalisations from a set of specific data.

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10.2.2 Examples of Machine-Based Learning

Inductive learning systems work on sets of example data from a process or artefact whose behaviour it is required to model and they produce generalised behavioural rules from this data. Examples of how inductive systems work are given below.

Table 10.1 below shows an example of the decisions made by a hypothetical HF FMS. The aim of the system is to decide which of a number of channels is capable of meeting a requirement to propagate to a wanted station 1 whilst not propagating to an unwanted station 2.

Table 10.1 Example Data for an Inductive System

Prop. to Sta. 1	Prop. to Sta. 2	Chan. No.	Diagnosis
yes	no	7	good
no	no	1	bad
yes	no	8	good
yes	yes	2	bad
no	yes	21	bad
no	no	38	bad
yes	yes	24	bad
yes	no	6	good
yes	no	3	good
yes	no	4	good

A human operator of the system would use the following criterion for selecting "good" channels:

> IF channel propagates to required station 1 AND channel does not propagate to the station 2 THEN channel is good ELSE channel is bad.

Using the data from Table 10.1 alone another rule for selecting good channels would be:

IF 3 < channel number < 9 THEN channel is good

It is just as likely that an inductive system would produce the above rule from this data as it would produce the other. In the context of an HF FMS the above rule is physically meaningless: the channel number is not directly related to the performance of the channel.

In other examples the production of rules by induction may yield a relationship that had been overlooked by the system designers. This can give valuable insight into the way a system operates. However, in most cases where it is required to replicate behaviour in order to automate a process, meaningful, domain-oriented rules are required. In order to ensure their production the criteria and attributes presented to an inductive learning system need to be selected with care.

The example data in Table 10.2 was fed into an inductive learning package (the data has been adapted from an example presented in (Hart, 1989)).

Table 10.2 Example Inductive Learning System Input

Chan	Prop	Noise	I/f	Sec	Diag
1	30	65	79	50	Ave
2	40	32	62	50	Ave
3	65	70	66	86	Exc
4	30	32	30	50	Poor
5	40	42	56	60	Good
6	50	60	77	64	Good
7	25	32	30	50	Poor
8	72	64	75	75	Exc
9	38	40	45	42	Good

The above table represents the inputs to the decision making algorithm of an FMS, where:

Chan : Channel number.

Prop : Propagation score for the channel and path as a percentage. The higher this value is the higher is the quality of the channel from a propagation point of view.

Noise : Natural noise score for the channel and path as a percentage. The quietest channels have the highest scores.

I/f : Interference score for channel as a percentage. The channels with the least interference have the highest scores.

Sec : Security score for channel as a percentage. The most secure channel has the highest score.

Diag : Diagnosis for channel. It can be "exc" (excellent), "good", "ave" (average), or "poor".

The rules used to diagnose the channels were as follows:

(i) IF any single attribute has a score > 40 THEN the channel is rated good for that attribute.

(ii) IF the average score is > 70 THEN the channel is diagnosed as being excellent.

(iii) IF all attributes are rated good THEN the channel is diagnosed as being good.

(iv) IF an attribute has a score which is >34 but <40 THEN the attribute is marginal.

(v) IF a channel has one marginal attribute THEN it is diagnosed as being good.

(vi) IF the number of attributes with a score < 40 is < 2 and > 0 THEN the channel is diagnosed as being average.

(vii) IF the scores of >2 attributes are < 40 THEN the channel is diagnosed as being poor.

An inductive learning package was fed the information in Table 10.2 and it induced the following rules:

(i) IF propagation score > 58 THEN the channel is diagnosed as excellent.

(ii) IF propagation score < 58 and the interference score is < 37 THEN channel is diagnosed as poor.

(iii) IF propagation score < 58 and interference score > 37 and security score < 55 and security score > 46 THEN channel is diagnosed as being average.

(iv) ELSE channel is diagnosed as being good.

The resulting rules were somewhat different in nature to the original ones. However they work for the limited set of data in Table 10.2 above.

The data in Table 10.2 was re-formatted to give that shown in Table 10.3 below.

Table 10.3 Revised Inductive Learning System Input

Chan	Attribs Good	Attribs Marginal	Ave	Diag
1	3	0	56	Ave
2	3	0	46	Ave
3	4	0	72	Exc
4	1	0	36	Poor
5	4	0	49	Good
6	4	0	65	Good
7	1	0	34	Poor
8	4	0	71	Exc
9	3	1	41	Good

The rules produced by the same induction system from this data were as follows:

(i) IF average score > 53 and average score < 68 and number of attributes good < 3 THEN channel diagnosed as average.

(ii) IF average score $>$ 53 and average score $<$ 68 and number of attributes good = 4 THEN channel diagnosed as good.

(iii) IF average score $<$ 53 and number of attributes good = 4 THEN channel diagnosed as good.

(iv) IF average score $<$ 53 and number of attributes good = 1 THEN channel diagnosed as poor.

(v) IF average score $<$ 53 and number of attributes good = 3 and number of marginals = 1 THEN channel diagnosed as good.

(vi) IF average score $<$ 53 and number of attributes good = 3 and number of marginals = 0 THEN channel diagnosed as average.

(vii) IF average score $>$ 71 THEN channel diagnosed as excellent.

In this case, adjusting the format of the information fed to the inductive learning package dramatically altered its output. The integer cut-off values in the rules were slightly different from the originals. This is due to the fact that the inductive learning package only had a limited set of examples to work on.

When using inductive learning packages to generate rules for an expert system, it is often necessary to manually edit the rules produced by them. Hence, if an HF FMS was to be implemented as an expert system then operational data from an existing, successful FMS could be fed into an inductive learning package. The resulting rules would be examined and edited before the system's performance was compared with an operational system.

The mechanisms used to achieve inductive learning are complex. Considerable programmer effort is required to produce an inductive-learning rule generator. Fortunately, there are available inductive shell programs which take example data as their input and produce rules to be fed into an expert system, as their output. Some examples of typical shell programs are given below.

10.2.3 Example Inductive Learning Expert System Shells

Two commercially available expert system/inductive learning shells have been examined: Extran 7.2 and Xi Plus/ Xi Rule. These are described below.

Extran 7.2 is produced by Intelligent Terminals Ltd., Glasgow. It is an expert system shell with an inductive rule generator, which takes example data as its input and produces rules in Fortran-77 as its output. It runs on a wide variety of hardware, from IBM mainframes down to IBM pc-AT's and it has been used to implement expert systems for a wide variety of applications, including the analysis of data from space-craft engine tests, electronic ciruit board fault diagnosis, design of gas-oil separators and the control of a urea processing plant. It can be configured to work either in an interactive, advisory mode or as an embedded system controller, taking its input from other software routines and processes. The shell costs £1995, for a pc-based version.

Xi Plus is an expert system shell, produced by Expertech Ltd, Slough. Xi Rule is the inductive rule generator which can be used in conjunction with Xi Plus. The current version of Xi Plus only works in an interactive fashion, but there is an embedded version under development which should be ready during the autumn of 1989. Xi Rule runs on minicomputers and it can produce its output rules in C so that they could be embedded within a user-written shell or Xi Plus, running on a pc. There are many, documented applications of Xi Plus in the engineering, manufacture, commerce, finance and retail sectors. The embedded version of Xi Plus will cost approximately £2000 and Xi Rule costs an additional £350.

10.3 Design of an HF Communications System Using Expert System and Inductive Learning Techniques

Expert system and inductive learning shells are readily available at a relatively low cost (especially when the saving in man-hours arising from their use is taken into consideration). As a result many organisations with no in-house AI expertise are able to use them for programming applications that would previously have required specialised personnel. This section of the thesis aims to show how an HF communications system controller could be produced using these techniques.

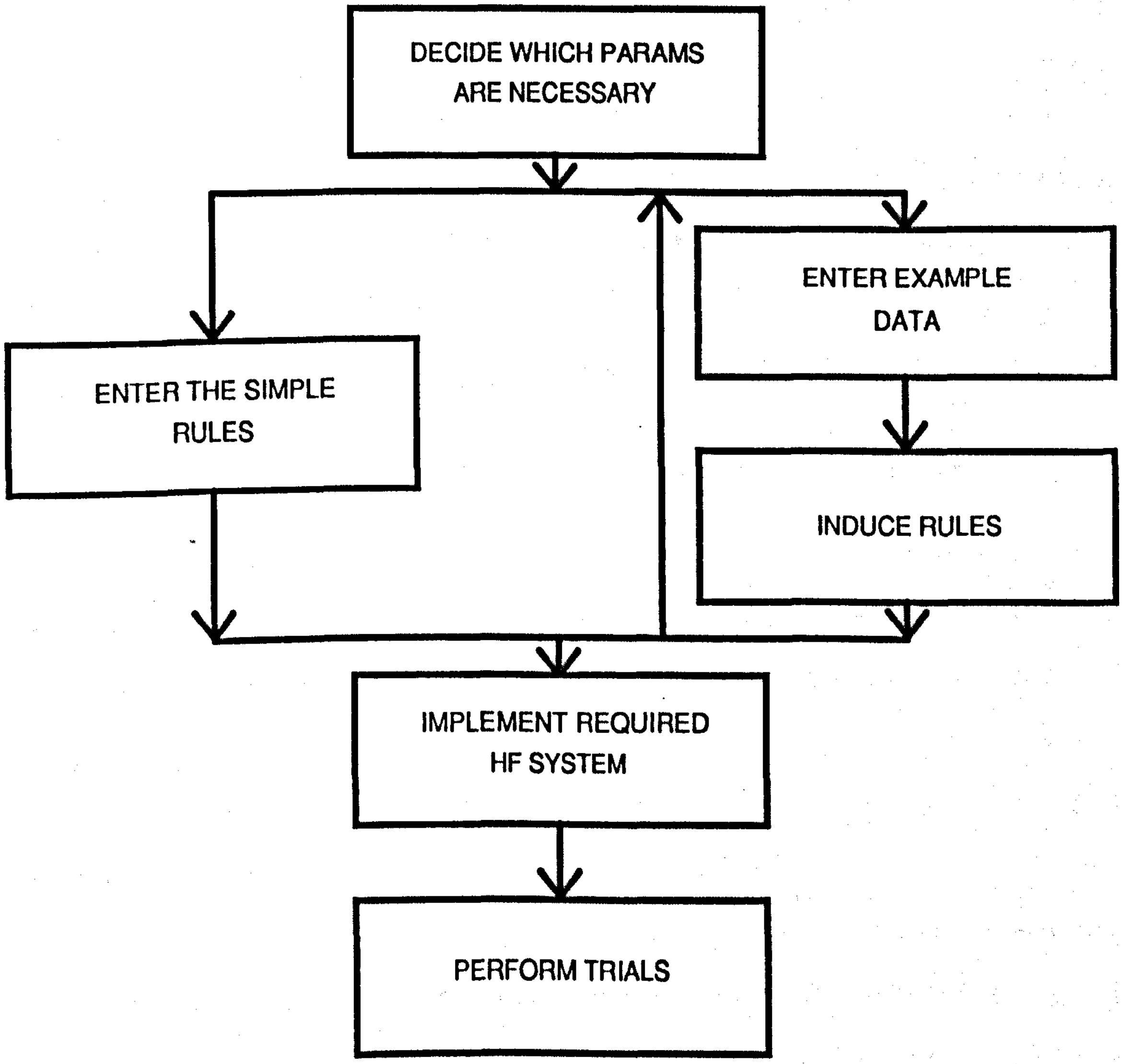
As has been previously described, induction systems generate rules from example operational data. The Plessey HF system has logged all the data resulting from its trials and this has been stored on computer tape. The proposed design methodology would use this data to induce rules about the way an HF system should behave. The aims of this work are as follows:

(i) To produce an efficient and effective HF communications system controller.

(ii) To reduce the size of the code necessary to implement an adaptive HF terminal, thus allowing a greater number of users to benefit from adaptive techniques (especially mobile users).

(iii) To investigate which operational parameters have most effect on the performance of HF communication systems.

FIGURE 10.1. HF SYSTEM DESIGN LIFE-CYCLE



A design life-cycle showing how it is proposed to produce an HF communication system controller using inductive methods is shown in Figure 10.1.

Initially, the relevant input parameters to the system, such as sun-spot number, interference levels etc would need to be identified. A set of basic rules would then be produced, eg IF channel does not propagate THEN do not use. The example data would be entered into the induction shell to produce the more complex behavioural rules. The resulting rule set would be examined and edited by the system's designers. The rule induction process could be carried out iteratively in order to minimise the number of input data items required by the system to make operational decisions. Minimising the number of data items in this way would in turn reduce the number of measurements required to be made by the system, thus increasing its availability for communication purposes.

Once a satisfactory rule set had been found a detailed systems architecture could then be produced. The system configuration would be dictated by the data requirements of the rules.

The system would be able to adapt itself to changes in channel conditions. Such perturbations would result in changes in the data input to the rules: provided that the rules were correct, the required change in the system's operational parameters would be made. A record could be kept of the performance of the system enabling new rules to be induced if required. The resulting system could be trialled against an existing automatic HF system to yield initial performance comparison figures.

It is recognised that the architecture proposed in this chapter is little more than a new concept. However, the experience gained with existing automatic HF system architectures and operation during the course of the work does suggest that the concept may be capable of opening up fundamentally new design and control methodologies which break out of the conventional HF mould.

11 FUTURE WORK AND CONCLUSIONS

11.1 Aims

The three main aims of this research programme were presented in Chapter 6:

(i) Enhancement of HF PAP performance.

(ii) Extraction of interference information and signal adaptation in response to this information.

(iii) Development of design methodologies for automatic HF communication systems.

The preceding four chapters describe the work carried out in each of the above areas. The following enhancements and additions to the techniques and proposals presented are suggested:

11.2 Enhancement of HF PAP Performance

The aim of this work was to develop accuracy enhancement routines for embedded HF PAP's. In order to achieve this aim a chirpsounder-based propagation measurement system was constructed. This was to be used in the provision of path parameters for PAP accuracy enhancement routines.

At present, the chirpsounder monitoring equipment is capable of estimating the MOF's and LOF's for the paths between a selected chirpsounder transmitter and the monitoring terminal. Detection is accomplished by means of a matched filter for a nominal 3 kHz section of the chirp. Amplitude information is discarded: detection operates on the basis of zero crossing information only. In order to enhance the performance of the chirp monitor, and to make use of the propagation data provided by it, it is proposed that the following areas should be investigated.

11.2.1 Receiver Bandwidth

The performance of the chirp detector could be improved by increasing the receiver bandwidth. This should enhance the selectivity of the chirp matched filter. If the selectivity was improved sufficiently then it may be that successful chirp detection could be accomplished with the filter taps and filter input samples represented in an analogue form rather than in the limited fashion implemented in the current chirp detector. This would allow the impulse response of a channel to be determined, thus enabling the multipath structure to be derived for a particular link. A "rough" ionogram could then be produced, as described in Chapter 7.

11.2.2 Interpolation of Chirpsounder Data

Procedures could be developed to enable selected chirpsounder data to be interpolated or extrapolated to provide estimated path parameters for a user's specific path, as mentioned in Chapter 7. This could include area coverage predictions for network scenarios.

11.2.3 Estimation of Path SNR

The strength of the received chirp signal could be used in conjunction with passively monitored interference levels, to provide the user with an estimated SNR for a path. This information could then be used either as RTCE data for an operational link or it could be used by PAP update and enhancement routines.

11.2.4 Filter Characteristics Compensation

As detailed in Chapter 7, band-pass filters can exhibit severe phase distortion at the edges of the pass-band. Since the chirp matched filter will be most sensitive to phase variations occuring at the high end of the pass-band, characterising the phase response of the IF filter in the receiver and adjusting the matched filter parameters accordingly, should improve its performance.

11.2.5 PAP Update Procedures

Procedures for enhancing the performance of embedded propagation models, using the data derived from chirpsounder monitoring, need to be developed. The techniques of sun-spot number manipulation and time and frequency shifting of measured and predicted MUF profiles could be investigated initially. An analysis of PAP input parameters could be performed in order to determine which are the most suitable for use in accuracy enhancement algorithms.

11.2.6 Terminal Chirpsounder Transmitter

Given the simplicity and low implementation costs offered by use of DSP devices, it may be advantageous to develop a communications systems' oriented chirpsounder transmitter. This need only scan the system's frequency allocations . The chirp transmitter would radiate in-band chirp segments on each allocation and a synchronised chirp receiver would then be used to derive the path parameters.

11.3 Extraction of Interference Information

There is currently a lack of adequate interference data input for automatic HF communication systems. The provision of such information is especially important in regions where levels of HF spectral occupancy are high. The aims of this part of the research programme were to provide interference trend information, pertaining to the system's frequency allocations, and to provide data concerning the in-band interference structure, in order to enable optimisation of signal formats.

Much effort was spent designing and testing template correlation algorithms, after which there was insufficient time remaining to develop channel interference trend extraction routines. Hence, it is suggested that their implementation development be part of a future research programme.

The principle of template correlation has been investigated theoretically and, to a certain extent, practically during this research programme. Further work is necessary, to examine its use as a tool to aid the placement of signal tones within a communications bandwidth. This will require the construction of a suitable FSK modem and bench and on-air trials to be carried out.

The technique of spectrum complementing, a more advanced form of tempate correlation, needs to be implemented to at least a simulation stage. A limited version of this technique could be used to optimise the transmission energy density spectrum of, for example, a multi-tone FSK modem, in order to minimise the BER.

It would seem appropriate that template correlation, in a practical form, should be incorporated into a modem architecure, implemented using a DSP device, eg TMS 320C25.

11.4 Development of Design Methodologies for Automatic HF Communication Systems

The aims of this work were to develop a structured and coherent design methodology and systems' architecture for automatic HF communications systems. The structure of the FMS, embedded within the overall system architecture, was to receive the most attention, ie ways in which the various FMS modules and routines can be combined in an effective manner.

Two system designs were produced. The ideas presented in Chapter 9 use conventional FMS tools and a structured architecture was developed around them. Chapter 10 describes an approach which is essentially a radical departure from conventional HF system design techniques.

The ideas presented in Chapters 9 and 10 require validation and implementation to at least a simulation level. The inductive learning-based system, described in Chapter 10, is a significant departure from current HF communication system design methodologies. It is felt that this terminal implementation warrants further study and refinement.

11.5 General Aims

It is proposed that the overall objective of any future work carried out on the topics described in this thesis should be to develop a compact HF terminal architecture, which incorporates the refined versions of the frequency management tools that have been developed, and which is compatible with both point-to-point and network deployment.

11.6 Conclusions

The programme of research described in this thesis has achieved the following:

(i) A compatible HF terminal architecture with structured and coherent software which is capable of incorporating different types of hardware units and software modules, as required.

(ii) A flexible chirpsounder monitoring unit with the ability to provide data for use in the enhancement of PAP accuracy.

(iii) A flexible and deterministic algorithm (template correlation) which facilitates the optimisation of signal formats in order to avoid in-band interference, minimise BER and maximise use of the capacity of an HF channel.

(iv) Two schematic designs for HF systems with fundamentally new control methodologies.

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13 Acknowledgements

I would like to thank the Science and Engineering Research Council for funding the research programme. Also, I am grateful for the funding and expert advice provided by members of RC Division, Plessey Research (Roke Manor) Ltd.

I wish to thank my colleagues and friends in the Department of Electronic Engineering at the University of Hull. In particular, the advice and guidance provided by my supervisor, Professor M. Darnell is appreciated. Thanks also to Mr. J. Hague and Dr. E.D. Chesmore for invaluable advice and discussion sessions.

The help provided by Mr A. Collinson, SERCO Ltd, RAF Fylingdales with all AI-related matters is acknowledged.

I wish to thank my family and close friends for encouragement and support.

Special thanks to Tanya Wood for her support and love.

14 Published Work

(i) October, 1987

"Artificial intelligence in HF communication systems"

Presented at AGARD Conf. On "Effects of Electromagnetic Noise and Interference on Performance of Military Radio Communication Systems", Lisbon, AGARD Conf. Publ. No. 420.

(ii) October, 1987

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IEE Colloquium Digest No. 1989/33

Three further publications on chirpsounder monitoring, template correlation and the design of HF communication systems are currently being prepared for submission to the IEE proceedings.